Administrator Guide

Yeastar P-Series Cloud Edition

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About This Guide

In this guide, we describe every detail on the functionality and configuration of the Yeastar P-Series Cloud Edition.

Audience

This guide is for administrators who need to prepare for, configure, and operate the PBX system. We begin by assuming that you are familiar with networking and other IT disciplines.

Getting Started

Activate Yeastar P-Series Cloud Edition

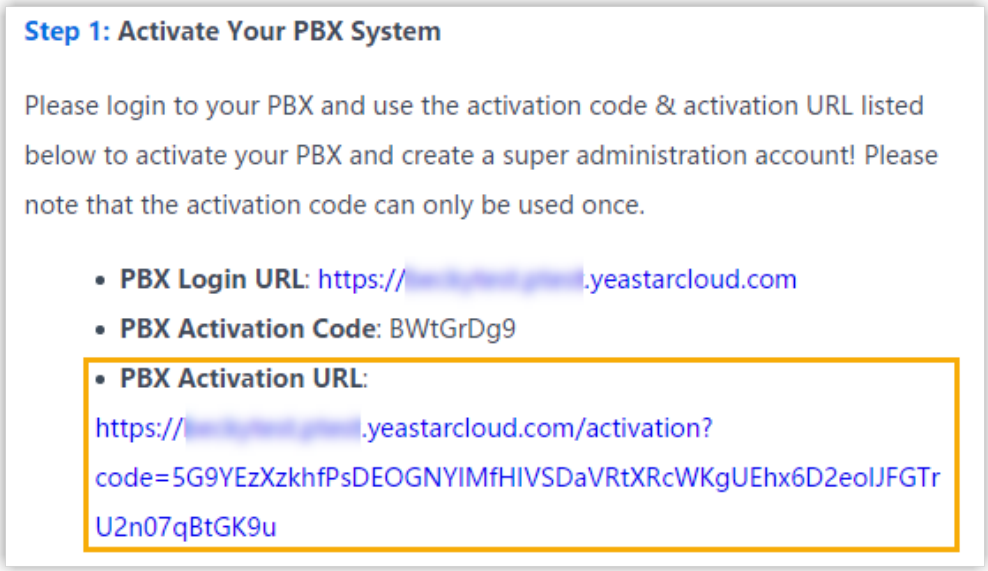
Before starting to use the Yeastar P-Series Cloud Edition, you need to activate the PBX first. The activation credentials and PBX login URL are provided in the welcome email of Yeastar P-Series Cloud Edition.

Prerequisites

You have received the welcome email of Yeastar P-Series Cloud Edition.

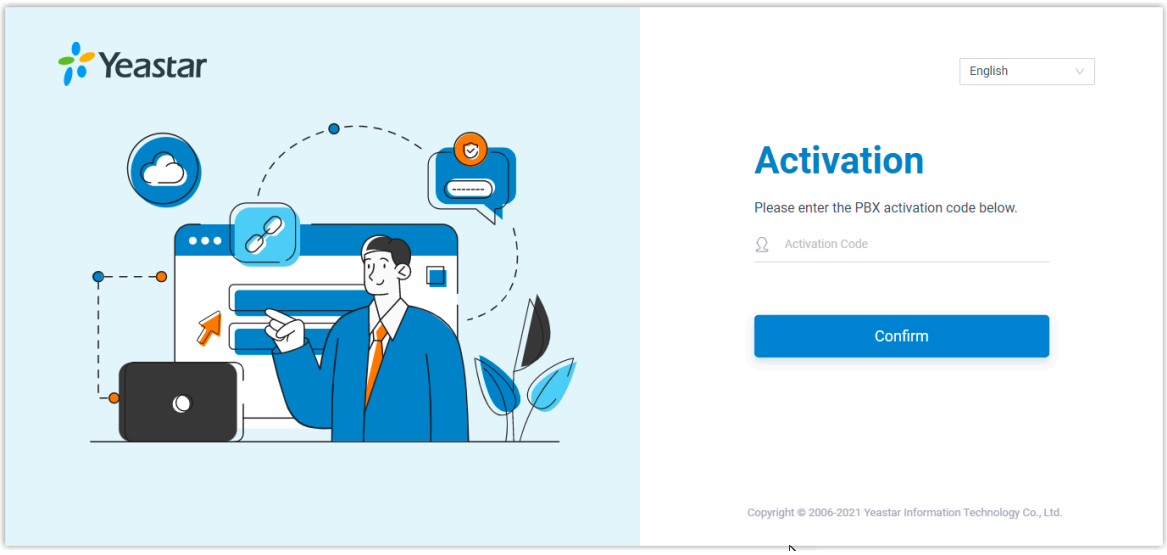
Procedure

1. In the welcome email, click the **PBX Activation URL** to enter the PBX activation page.



2. In the **Activation Code** field, enter the PBX activation code provided in the welcome mail.

> **Note:**
> The PBX activation code can only be used once.



3. Click **Confirm**.

Result

You are redirected to the Installation Wizard. For more information, see Initial Setup Using the Installation Wizard.

> **Important:**
> Only when you finish the system initial configurations can the Yeastar P-Series Cloud Edition be activated successfully.

Initial Setup Using the Installation Wizard

When you access the PBX web portal for the first time, you need to finish initial configurations for the system using the Installation Wizard.

Prerequisites

You have accessed the PBX web portal and entered the Installation Wizard.

For more information about how to access the PBX web interface, see Log in to PBX Web Portal.

> **Warning:**

> The Installation Wizard only appears when you first configure the system with factory settings.

Procedure

- Step 1. Set up super administrator account
- Step 2. Configure the system time
- Step 3. Localize and customize the system
- Step 4. Check and confirm the configurations

Step 1. Set up super administrator account

1. In the **Basic** section, enter the information of the super administrator account.

   > **Note:**
   > - Do NOT forget the username and password of the super administrator account, or you need to reset your system to reconfigure the account and log in to the PBX.
   > - The super administrator has access to all features on the system, and the super administrator can assign administrator role to users. For more information, see User Roles and Permissions.

   - **Username**: Specify the username that is used to log in to PBX web portal.
   - **Password**: Specify the password that is used to log in to PBX web portal.
   - **Repeat the password**: Repeat the password to confirm.
   - **Email Address**: Enter the email address of the super administrator.

     The email address can be used to receive system notifications, reset web login password, and log in to the administrator portal.
   - **Mobile Number**: Enter the mobile number that can be used to receive system notifications.
   - **Prefix**: Optional. Enter the prefix according to the dial pattern of the outbound route, so that the system can successfully send calls to the mobile number.

     For more information of the prefix setting, see Prefix and Dial Pattern.
2. In the **Event Notifications** section, configure event notifications for the super administrator.
   - **Send Event Notification to PBX Administrator**: Decide whether to enable notifications for the super administrator or not.

- **Contact Name**: Enter the name of the super administrator.

  > **Note:**
  > This name helps you identify the super administrator from the Notification Contacts list.

- **Notification Level**: System notifications are divided into different levels according to importance. You can select notification levels to filter and receive the relevant notifications.
- **Notification Method**: Select method(s) to receive notifications.

For more information of event notifications, see Event Notification Overview.

3. Click **Next**.

Step 2. Configure the system time

1. In the **Date and Time** section, configure the time zone and daylight saving time.
2. In the **Display Format** section, select the display format for date and time.
3. Click **Next**.

Step 3. Localize and customize the system

1. In the **System Prompt Language** section, select the radio button beside a system prompt to set it as the default system prompt.

   > **Note:**
   > Click **Download Online Prompts** to download more prompts.

2. In the **Other Settings**section, adjust the following settings for your local installation.
   - **Notification Email Language**: Select which language of email contents to be received.
   - **Device Name**: Specify a name for the PBX system.
   - **Name Display Format**: Select the display format for Extension User's Name and Contact Name.
   - **Tone Region**: Select your country/region or the nearest neighboring country/region to enable the default dial tone, busy tone, ring tone for your region.
   - **Enable Allowed Country/Region Code Dialing Protection**: To restrict users from making international calls, enable this option. When enabled, users can not make international calls to any countries or regions.

> **Note:**
> To allow users to make international calls to specific countries or regions, you need to grant permission to desired users, and set the allowed countries or regions. For more information, see Restrict International Calls to Specific Countries or Regions.

- **International Dialing Code**: Enter the prefix of international call according to your country.

  When a user tries to call a number starting with the prefix, the PBX's outbound route will identify this call as an international call.

3. Click **Next** to see the summary.

Step 4. Check and confirm the configurations

1. Check the all the configured settings on the **Summary** page.
2. To edit the configurations of a specific step, click beside the step title.
3. To edit the configurations of the previous step, click **Re-configure**.
4. If all the configurations are confirmed, click **Reboot** to take effect.

Result

All the configurations take effect after the system reboots.

You need to access the domain name of the PBX and log in to PBX web portal by the super administrator username and password.

Log in to PBX Web Portal

Yeastar P-Series Cloud Edition provides two different web portals for users with different roles to quickly set up and manage the system. This topic describes the difference between them, and introduces how to log in to the PBX web portal.

Web portals overview

Table 1.

Web portal	Description
Administrator portal	Dedicated web portal for super administrator.

Table 1. (continued)

Web portal	Description
	Super administrator has the highest privileges. Once logged in, the super administrator can access and manage all the PBX system features, including creating extension accounts for users and granting privileges to the created user accounts. **Login address**: `PBX domain name/admin`. For more information, see Log in to administrator portal.
Management portal	The web portal for users with administrative privileges. Users who have a specific role assigned by the super administrator can log in through this portal. Once logged in, users can only access and manage the specific PBX system features that are granted to their roles. **Login address**: `PBX domain name`. For more information, see Log in to management portal.

Log in to administrator portal

Prerequisite

An operation and maintenance terminal (a PC) is available. The PC must meet the following requirements:

- Have a web browser installed. The following table shows the compatible browsers.

  Table 2.

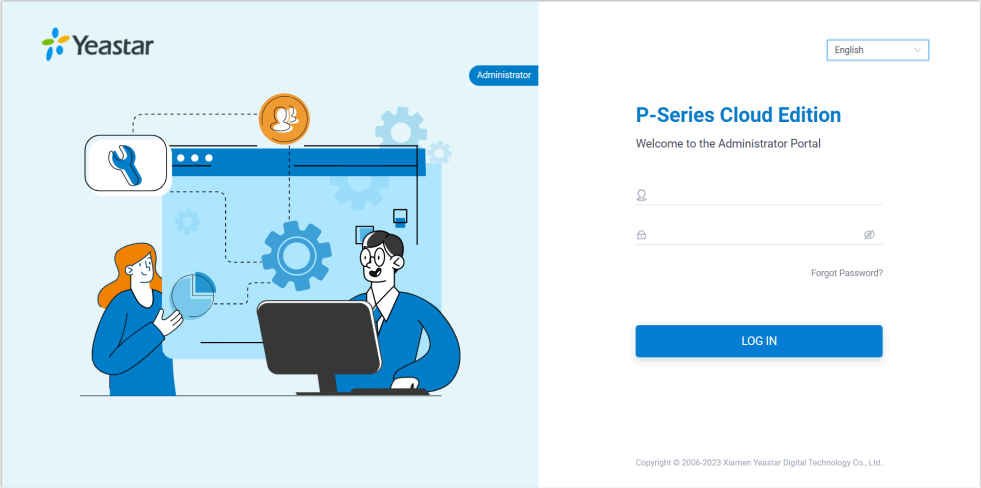
Web Browser	Version
Google Chrome (recommended)	Chrome 87 or later
Microsoft Edge	Edge 87 or later
Opera	Opera 72 or later

- Support the resolution of 1366 x 768 or higher.

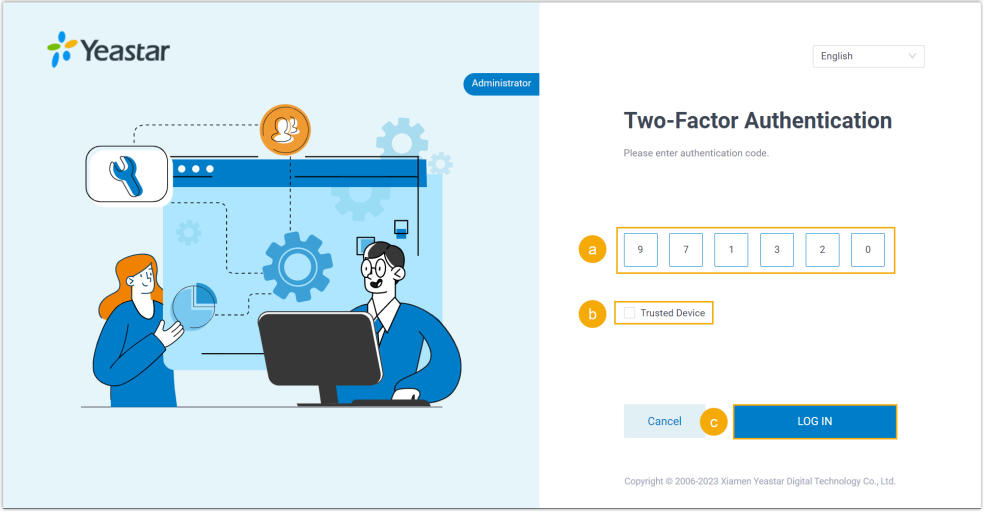
Procedure

1. Open the web browser, enter the domain name in the address bar, followed by a forward slash and the word "admin", i.e. **PBX domain name/admin**, and press **Enter**.

2. Enter the credentials of super administrator account, then click **LOG IN**.
   - **Username**: The **username** or **email address** of super administrator account that you have configured in the Installation Wizard.
   - **Password**: The password of the super administrator account.



3. If you have set up two-factor authentication, you need to enter an authentication code.



   a. Enter the authentication code provided by an authenticator application or email.
   b. **Optional:** Select the checkbox of **Trusted Device**.

> **Note:**
> For the device from which you log in most frequently, you can select the option to add it as a trusted device. In this

way, you don't have to re-enter an authentication code with this device for the next 180 days.

c. Click **LOG IN**.

Log in to management portal

Prerequisite

An operation and maintenance terminal (a PC) is available. The PC must meet the following requirements:

- Have a web browser installed. The following table shows the compatible browsers.

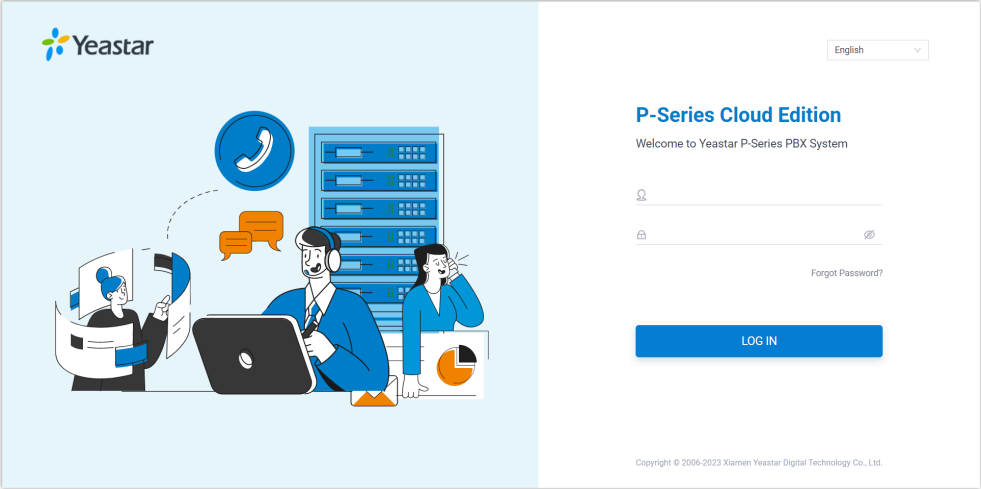
Table 3.

Web Browser	Version
Google Chrome (recommended)	Chrome 87 or later
Microsoft Edge	Edge 87 or later
Opera	Opera 72 or later

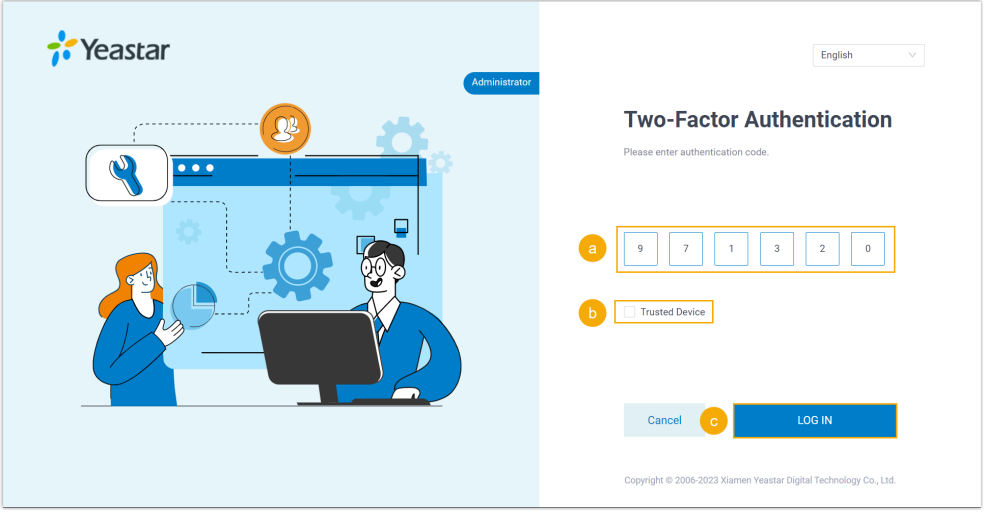
- Support the resolution of 1366 x 768 or higher.

Procedure

1. Open a web browser, enter the domain name of your PBX in the address bar, and press **Enter**.
2. Enter the credential of the user account, click **LOG IN**.
   - **Username**: The **email address** of the user account.
   - **Password**: The password of the user account.



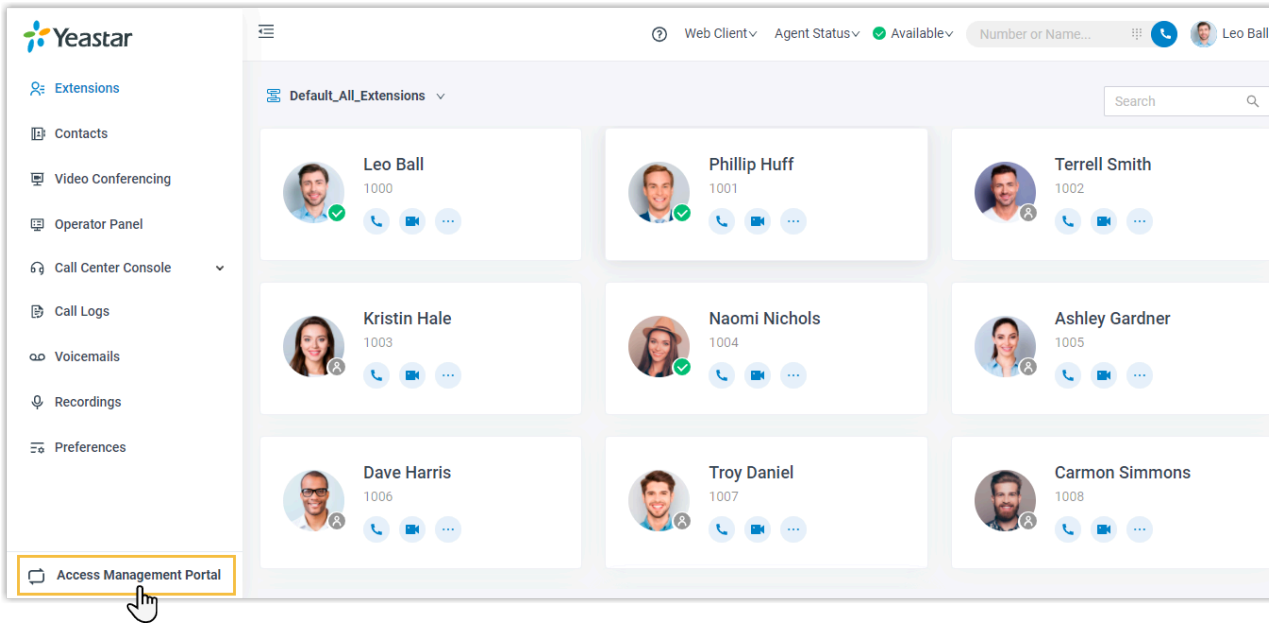
3. If you have set up two-factor authentication, you need to enter an authentication code.



   a. Enter the authentication code provided by an authenticator application or email.
   b. **Optional:** Select the checkbox of **Trusted Device**.

      > **Note:**
      > For the device from which you log in most frequently, you can select the option to add it as a trusted device. In this way, you don't have to re-enter an authentication code with this device for the next 180 days.

   c. Click **LOG IN**.
4. In the bottom left corner, click **Access Management Portal**.



Change the Password of Super Administrator

If you know the current password of super administrator, you can log in to the PBX administrator portal and follow the steps to change the super administrator's password.

Background information

The username and password of super administrator are configured in Installation Wizard.

Important:

- The username of super administrator cannot be changed unless your reset the system.
- If you forget the password of super administrator, you can reset the password. For more information, see Reset the Password of Super Administrator.

Procedure

1. Log in to PBX administrator portal.
2. At the top-right corner of the web page, click and select **Change Password**.

3. On the pop-up window, enter the old password and new password.
4. Click **Save**.

Result

The password is reset, you will be logged out of the web page automatically. To log in to PBX administrator portal, enter the new password.

Reset the Password of Super Administrator

As a super administrator, you can reset your web login password if you forget the password.

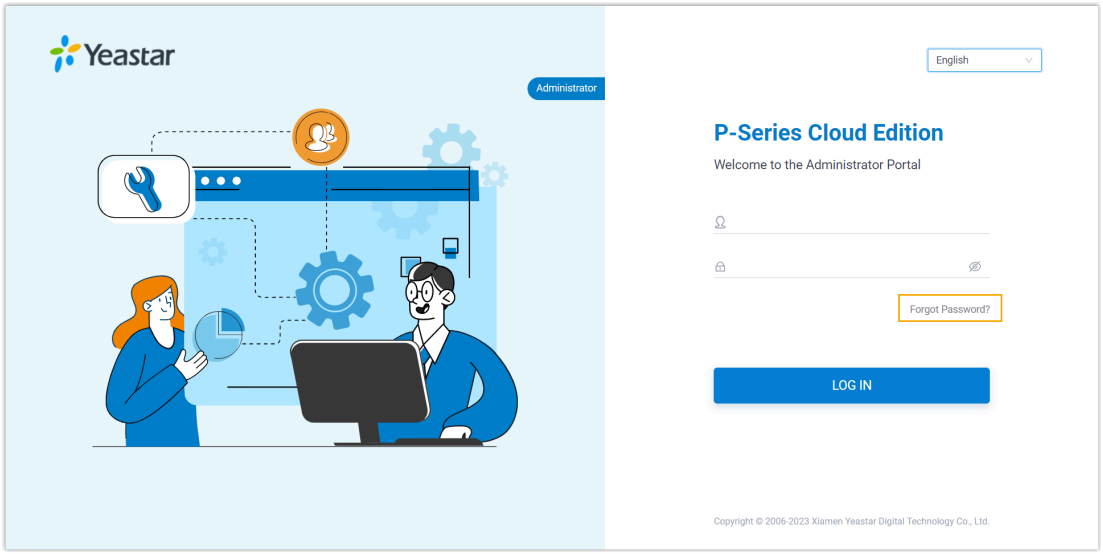
Prerequisites

- You need to provide both username and email address, or you cannot reset your password.

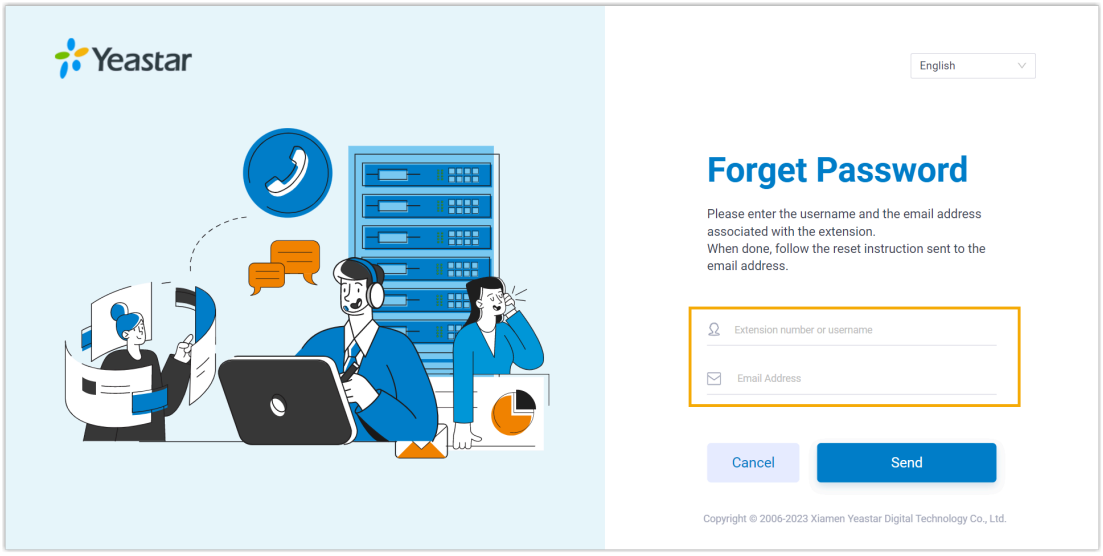
> **Important:**
> If you forget the username of super administrator, you need to reset the system to reconfigure a new username.

Procedure

1. Access the PBX web login page, click **Forgot Password?** to enter the **Forget Password** page.



2. On the **Forget Password** page, enter the following information:
   - **Extension number or username**: The username of super administrator.
   - **Email Address**: The email address that is associated with the super administrator.



3. Click **Send**.

   A password reset email is sent to super administrator's email address.
4. Check the password reset email, and click the link provided in the email to enter the **Reset Password** page.

Note:

> This link is valid for 30 minutes and can only be used once.

5. On the **Reset Password** page, enter your new password twice, and click **Save**.

Result

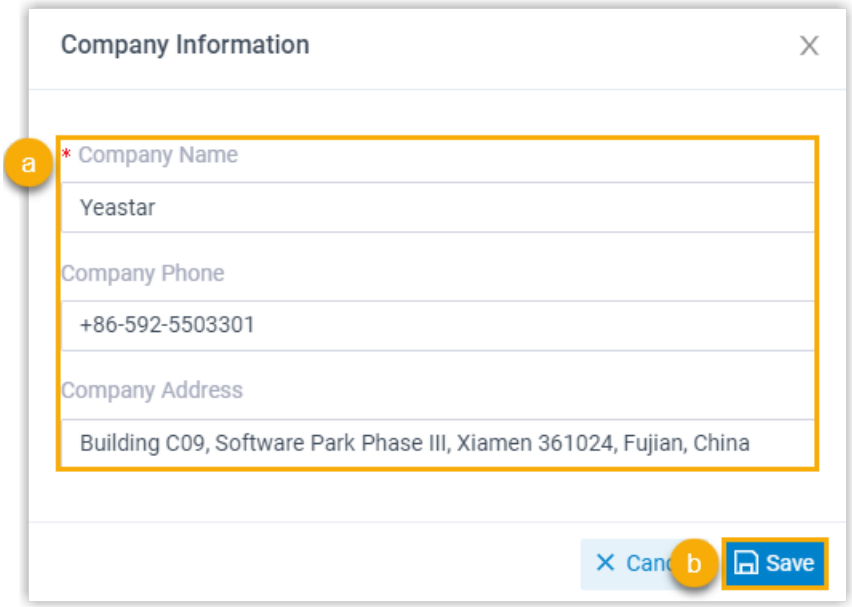
The password of super administrator is changed. You need to log in to PBX administrator portal by the new password next time.

Set up Company Information

Company information contains basic information about your company, including company name, company phone number, and company address. This topic describes how to set up company information.

Procedure

1. Log in to PBX web portal.
2. At the top-right corner of the web page, click and select **Company Information**.
3. In the pop-up window, do as follows:



   a. Configure the name, the phone number, and the address of your company as needed.

> **Note:**
> If you enable **Organization Management** feature on the system, the **Company Name** is required and will be used as the root organization name. For more information, see Enable or Disable Organization Management.

b. Click **Save**.

View System Information

This topic describes how to view a summary of information about your system firmware and network.

Procedure

1. Log in to PBX web portal, go to **Dashboard**.
2. At the top-right corner of **Dashboard**, click **Information**.



The following information is displayed:

- Device Name
- Product Model
- Serial Number
- Firmware Version
- System Time
- Uptime
- Maximum Extensions
- Maximum Concurrent Calls

Change Web Interface Language

The default web interface language of Yeastar P-Series Cloud Edition is English, the interface can be easily switched to the language of your choice.

Procedure

1. Log in to PBX web portal.
2. At the top-right conner of the web page, click  .
3. Select **Language** and select your desired language.

   The web interface is switched to the selected language immediately.

Log out of PBX web portal

When you're ready to quit the Yeastar P-Series Cloud Edition, simply close the web page or follow the steps below to log out of the PBX web portal.

Procedure

1. At the top-right conner of the web page, click  .
2. Select **Log out**.

Dashboard

Dashboard Overview

Yeastar P-Series Cloud Edition Dashboard gives you a historical and real-time view of what is happening on the PBX. This topic describes all the widgets on the Dashboard.

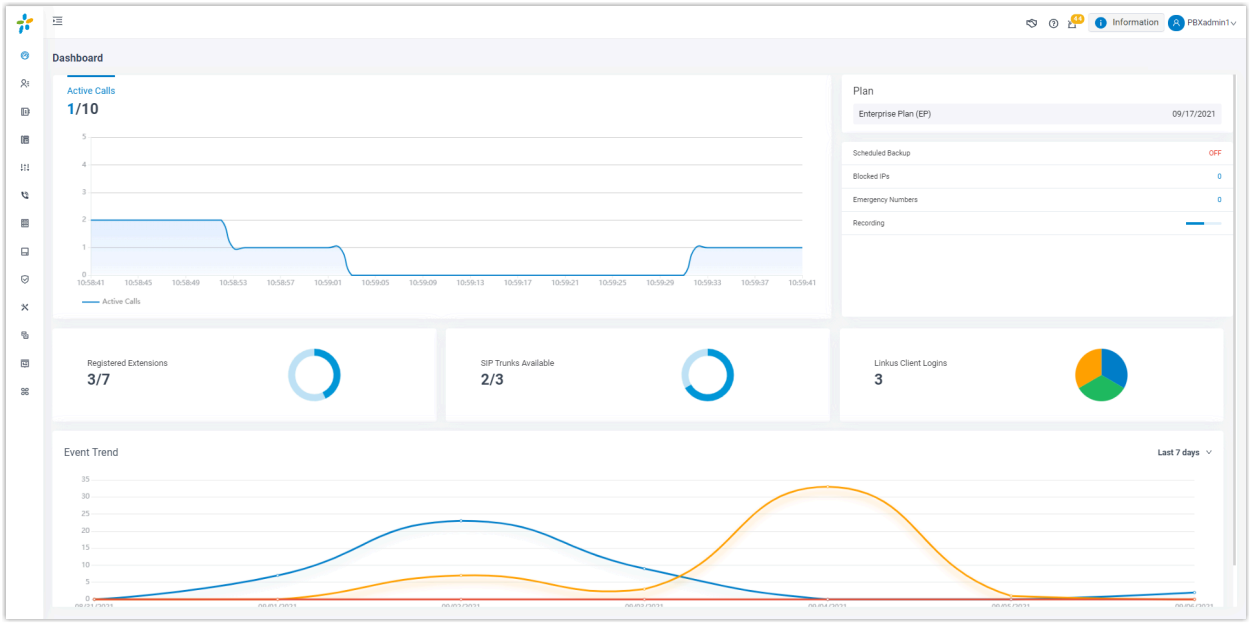
Yeastar P-Series Cloud Edition Dashboard provides widgets to help you monitor system performance in real time, and allows you to quickly access specific PBX features by simple click on headings.

The supported widgets are as follows:

- System performance
- System information
- Plan
- System status
- Event trend

1. System performance
2. System information
3. Plan
4. #unique_17_Connect_42_section_t34_2t3_nyb
5. #unique_17_Connect_42_section_lfp_lw3_mmb
6. System status
7. Event trend

1. System performance
2. System information
3. Plan
4. #unique_17_Connect_42_section_t34_2t3_nyb
5. #unique_17_Connect_42_section_lfp_lw3_mmb
6. System status
7. Event trend

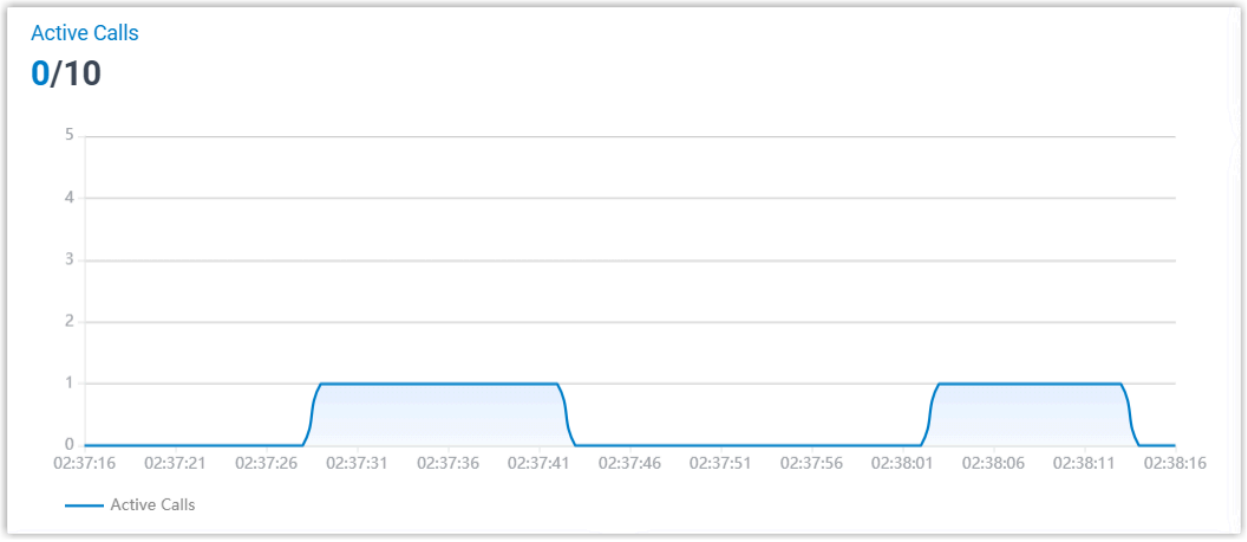


1. System performance
2. System information
3. Plan
4. System status
5. System status
6. Event trend

System performance

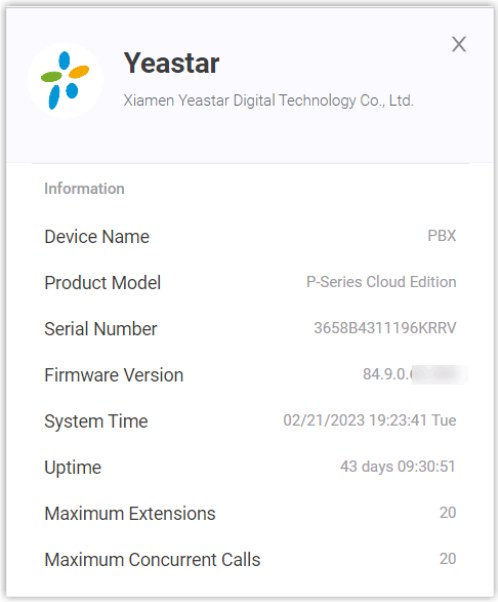
System performance displays the following information:

- **Active Calls**: The real-time and the supported concurrent calls.



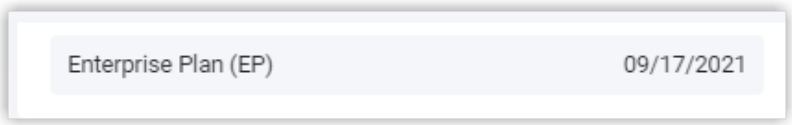
System information

Click **Information** at the top-right corner. System information displays the PBX's basic information.



Plan

Plan displays the plan that you subscribe to or try and its expiration date.



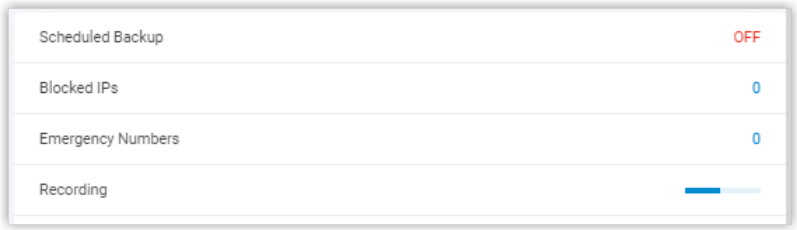
System status

System status displays the following information:

- **Registered Extensions**: The number of registered extensions and created extensions.
- **SIP Trunks Available**: The number of available trunks and created trunks.
- **Linkus Client Logins**: The number of Linkus clients where users has logged.

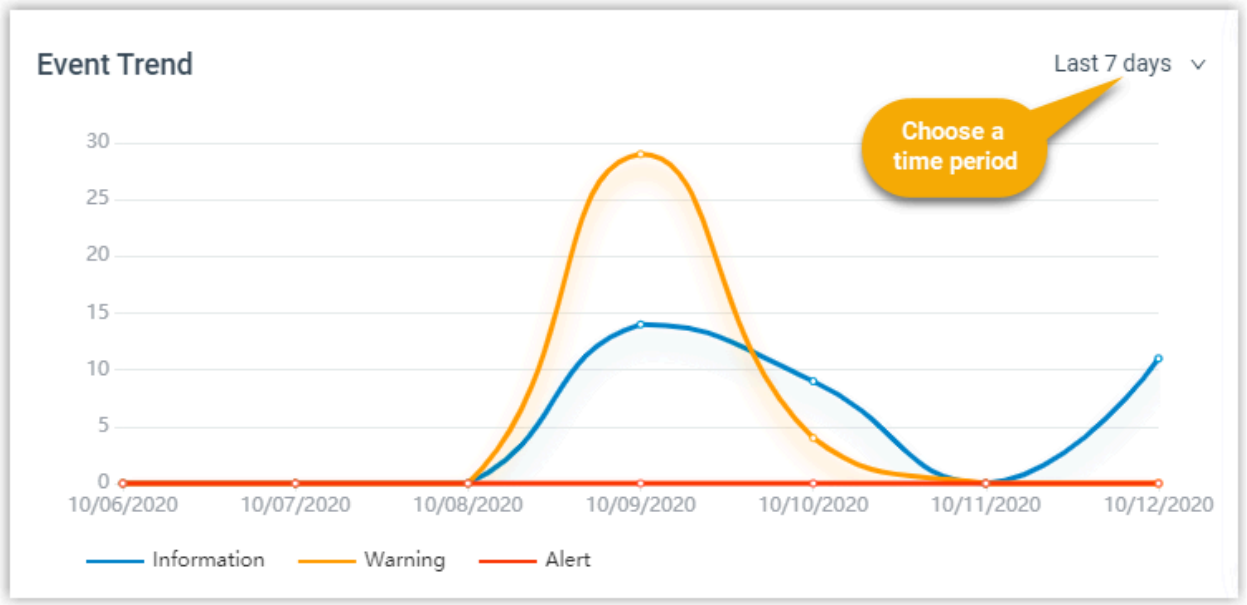


- **Scheduled Backup**: Whether scheduled backup feature is enabled or not. If enabled, the system displays the last time when a backup file was created.
- **Blocked IPs**: Display the following information:
  - The number of IP address and account that were blocked by the PBX.
  - The last time when an IP address or an account was blocked by the PBX.
- **Emergency Numbers**: The number of created emergency numbers.
- **Recording**: How much storage space for recording has been used.



Event trend

Event trend provides historical and real-time view of system events. You can track frequency of events that were triggered during the last 7 days, 15 days, or 30 days.


Event Trend
Last 7 days
Choose a time period
30
25
20
15
10
5
0
10/06/2020
10/07/2020
10/08/2020
10/09/2020
10/10/2020
10/11/2020
10/12/2020
Information
Warning
Alert

Extension

Extension Overview

An extension is a short internal number. Extensions allow users to make and receive calls. You can assign extensions to every employee in your organization.

Extension types

Yeastar P-Series Cloud Edition supports SIP extension, which is based on SIP protocol.

To use a SIP extension to make or receive calls, you need to register the extension on an IP phone or a softphone.

For more information, see Create a SIP Extension and Set up a SIP Phone.

Online status

Online status allows you to view status of phone endpoints and Linkus clients.

- **Phone endpoints**
  -  indicates that the SIP extension is registered and ready for use.

    Hover your mouse over  to view the IP addresses of SIP phones where the extension is registered.

- **Linkus clients**
  -  indicates that Linkus Desktop Client is ready for use.
  -  indicates that Linkus Mobile Client is ready for use.
  -  indicates that Linkus Web Client is ready for use.

Create Extensions

Create a SIP Extension

This topic describes how to create a SIP extension and configure relevant settings.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click **Add** and select **Add**.
2. In the **User Information** section, configure user information as follows:
   - **First Name**: Enter the user's first name.
   - **Last Name**: Enter the user's last name.
   - **Email Address**: Enter the user's email address. The user can log in to PBX web portal and Linkus clients, receive voicemail messages and PBX notifications via the email address.

     > **Note:**
     > An email address is exclusive to a user.

   - **Mobile Number**: Enter the user's mobile number. The user can receive calls or PBX notifications on this mobile number.
   - **User Password**: Enter a user password. The user can use the password to log in to Linkus clients.

     > **Note:**
     > The password is randomly generated by default. To change user password, a minimum of 10 characters with number, upper case, and lower case are required.

   - **User Role**: Assign a role to the user to determine whether the user can manage specific PBX features.

     The default value is **None**, which means that the user can not manage specific PBX features.

     > **Note:**
     > The system has default user roles with pre-configured permissions. You can also Create a User Role.

   - **Organization**: Select one or more organizations to which the extension belongs.

     > **Note:**

> This option is available only when you enable the **Organization Management** feature.

- **Job Title**: Enter a job title for the user, which will be displayed on Linkus clients.

3. In the **Extension Information** section, configure extension information as follows:
   - **Extension Number**: Enter an extension number.
   - **Caller ID**: Enter a caller ID number. The caller ID will be displayed on the callee's device.
   - **Registration Name**: Enter a name that is used to register the SIP extension. The default registration name is randomly generated.
   - **Registration Password**: Enter a password that is used to register the SIP extension. The default registration password is randomly generated.

     > **Note:**
     > For security reasons, we recommend that you set a strong password.

   - **IP Phone Concurrent Registrations**: Select a value from the drop-down list.

     This option defines how many SIP endpoints are allowed to register with the extension.

     > **Note:**
     > - The maximum number of concurrent registration is 3.
     > - Concurrent Registration setting only limits the registration number of non-Linkus SIP endpoints. The registration number of Linkus clients is not counted.

4. **Optional:** Click other tabs to configure other settings according to your needs.
5. Click **Save** and **Apply**.

Result

The SIP extension is created.

What to do next

- To set up a SIP phone, see Set up a SIP Phone.

Bulk Create SIP Extensions

This topic describes how to bulk create SIP extensions.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click **Add** and select **Bulk Add**.
2. Configure basic settings for the extensions as follows.
   a. In the **User Information** section, configure user information as follows:
      - **Start Extension Number**: Enter the start extension number.

        The system will bulk create extensions starting with the extension number.
      - **Create Number**: Enter the number of extensions that will be created.

        > **Note:**
        > Only an integer ranging from 1 to 999 is allowed.

      - **User Role**: Assign a role to the extensions to determine whether these users can manage specific PBX features.

        The default value is **None**, which means that these users can not manage specific PBX features.

        > **Note:**
        > The system has default user roles with pre-configured permissions. You can also Create a User Role.

      - **Organization**: Select one or more organizations to which the extensions belong.

        > **Note:**
        > This option is available only when you enable the **Organization Management** feature.

      - **Job Title**: Enter a job title for the extensions, which will be displayed on Linkus clients.

   b. In the **Extension Information** section, configure extension registration information as follows.
      - **Registration Name**: Choose how to configure registration name.
        - **Generate Randomly**: Registration name will be randomly generated for each extension.

- **Prefix + Extension Number**: If you choose the type, enter a prefix in the **Name Prefix** field.
- **Extension Number + Suffix**: If you choose the type, enter a suffix in the **Name Suffix** field.
- **Fixed Name**: If you choose the type, enter a fixed name in the **Fixed Name** field.
- **Extension Number**: If you choose the type, extension number will be the registration name of each extension.

- **IP Phone Concurrent Registrations**: Select a value from the drop-down list. This option defines how many SIP phones are allowed to register with each extension.

> **Note:**
> - The maximum number of concurrent registration is 3.
> - Concurrent Registration setting only limits the registration number of non-Linkus SIP endpoints. The registration number of Linkus clients is not counted.

3. **Optional:** Click other tabs to configure other settings for the extensions.
   - **Presence**: Configure presence settings.
   - **Voicemail**: Turn on **Enable Voicemail**, choose a password type from the drop-down list of **Voicemail PIN Authentication**.

> **Tip:**
> Configure voicemail notifications and play options according to your needs.

- **Generate Randomly**: A PIN code will be randomly generated for each extension.
- **Prefix + Extension Number**: If you choose the type, enter a prefix in the **PIN Prefix** field.
- **Extension Number + Suffix**: If you choose the type, enter a suffix in the **PIN Suffix** field.
- **Fixed Password**: If you choose the type, enter a PIN code in the **Fixed PIN Code** field.
- **Extension Number**: If you choose the type, extension number will be set to PIN code for each extension.

    - **Disabled**: No PIN code is required when accessing voicemails.
  - **Features**: Configure email notifications, time-conditional presence auto switch, call handling rules, call recording, etc.
  - **Advanced**: Configure advanced settings.
  - **Security**: Configure SIP security settings and call restriction settings.
  - **Linkus Clients**: Enable Linkus clients for the extensions.
  - **Function Keys**: Provision function keys.

    When the extensions are bound with phones through auto provisioning, the function keys associated with the extensions will be applied to phones.
4. Click **Save** and **Apply**.

Result

- The extensions are created.
- The registration passwords and user passwords of these extensions are generated randomly.
- The system prompts you the number of created extensions, and the associated extension numbers.

What to do next

- To set up a SIP phone, see Set up a SIP Phone.

Set up Phones

Set up a SIP Phone

This topic describes how to register a SIP extension on a SIP phone.

Prerequisites

- You have created a SIP extension.

Procedure

1. Gather information of extension registration.

For most SIP phones, the following credentials are needed in order to register with Yeastar P-Series Cloud Edition.

- The domain name of PBX
- SIP registration port: The default port is 5060 on Yeastar P-Series Cloud Edition.
- Transport protocol (Path: **Extension and Trunk > Extension > Advanced > Transport**)
- Extension information (Path: **Extension and Trunk > Extension > User**):
  - Extension number
  - Registration name
  - Registration password
  - Caller ID name

2. Register the extension on a phone.

   Log in to the phone's web interface, fill in and save the required items to register the SIP extension.
3. Confirm the extension's registration status in one of the following ways:
   - On the phone's web interface, check if the extension is registered.
   - Log in to PBX web portal, go to **Extension and Trunk > Extension**, check if the endpoint icon displays  in the **Online Status** column.

Result

The SIP phone is ready for use. Users can use the SIP phone to make and receive calls.

Extension Presence

Extension Presence Overview

This topic describes what is extension presence and how presence benefits a user's work.

What is presence

Presence indicates a user's current status. By default, anyone in your organization using Yeastar P-Series Cloud Edition can see if other users are available.

Yeastar P-Series Cloud Edition supports the following status:

- **Available**: The user is online and ready for communication.
- **Away**: The user is away from desk.

- **Business Trip**: The user is on a business trip.
- **Do Not Disturb**: The user doesn't want to be disturbed, and he or she won't receive any calls.
- **Lunch Break**: The user is currently on lunch break.
- **Off Work**: The user is currently off work.

How presence benefits a user's work

Presence is associated with the following settings. You can configure the following settings for each presence. When a user's presence changes, the following settings will change accordingly.

- **Presence information**: Details about current presence.
- **Call forwarding**: Route internal and external calls to different destinations based on extension presence.
- **Ring strategy**: Adjust endpoints' ring strategy based on extension presence.
- **Ring timeout**: Adjust endpoints' ring timeout based on extension presence except **Do Not Disturb** status.
- **Ring the Mobile Number Simultaneously**: Whether to simultaneously ring mobile phone when a call reaches the extension number.
- **Accept push notifications**: Whether to receive Linkus push notifications on Linkus Mobile Client, such as missed calls, voicemails, etc.
- **Agent Status Auto Switch**: Adjust agent status automatically if the user is in a queue.
- **Voicemail greetings**: Adjust voicemail greetings based on extension presence.

For more information, see Presence Settings and Change Voicemail Greetings.

Presence switch

There are two ways to switch extension presence:

- **Switch presence manually**: Extension users can switch their own presence on Linkus clients or by dialing a feature code; an administrator can also switch extension presence for specific users on PBX management portal.

  For more information, see Switch Presence on Linkus Client and Manually Switch Extension Presence.
- **Switch presence automatically**: Presence is switched based on Business Hours and Holidays.

  For more information, see Auto Switch Presence Status based on Business Hours and Holidays.

Presence Settings

This topic describes presence settings.

Background information

Yeastar P-Series Cloud Edition supports to configure presence settings under each presence for all the users. When a user's presence changes, presence settings will change accordingly.

Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension in the **Presence** tab.

- Presence Information
- Call Forwarding
- Ring Strategy
- Ring Timeout
- Options

Presence Information

Setting	Description
Presence Information	Add a note to the current presence. The note will be displayed on Linkus clients.

Call Forwarding

Call forwarding rules help forward incoming calls to a specific destination when the user is unavailable. You can set different destinations for incoming calls based on extension presence.

Setting	Description
Types of incoming calls	• **Internal Calls**: Set a call forwarding rule for incoming calls from colleagues. • **External Calls**: Set a call forwarding rule for incoming calls from external users.
Forwarding condition	Select a forwarding condition and configure a destination. • **Always**: Forward all incoming calls to the designated destination. • **No Answer**: Only forward unanswered calls to the designated destination.

Setting	Description
	• **When Busy**: Only forward the calls that come in while the user is talking on the phone to the designated destination.

Ring Strategy

Ring strategy allows you to decide in which order incoming calls are distributed to the end-points where the user's extension is registered.

- **Extension Endpoint**: The IP phone or softphone to which the user's extension has registered.
- **Linkus Mobile Client**
- **Linkus Desktop Client (Softphone Only)**
- **Linkus Web Client (Web Client Mode Only)**
- **Linkus Pad Client (SDK)**

> **Note:**
> - This option is ONLY available when Linkus SDK is enabled (Path: **Integrations > Linkus SDK**).
> - To enable users receiving incoming calls through Linkus Pad Client, you need to integrate the Linkus SDK for Android or Linkus SDK for iOS.

Setting	Description
Ring First	Set which endpoint will ring first.
Ring Secondly	Set which endpoint will ring secondly.

> **Note:**
> The ring strategy setting does not take effect if the extension is a member of a queue or a ring group. In such cases, when the extension receives incoming calls, all endpoints with the extension registered would ring simultaneously.

Ring Timeout

To prevent callers from waiting for a long time, you can configure ring timeout. If the call is not answered during the time period, it will be routed to the destination of **No Answer**.

Setting	Description
Ring Timeout	Enter a value or select a value from the drop-down list. **Note:** The valid range is from 5 to 300.

Options

Ring the Mobile Number Simultaneously

To simultaneously ring both extension and the associated mobile number when anyone calls in the extension number, you can configure a simultaneous ring strategy.

Note:
The feature is unavailable in **Do Not Disturb** status.

Setting	Description
Ring the Mobile Number Simultaneously	Check the option to enable this feature, and configure the user's mobile number.
Prefix	Enter the prefix of outbound route so that PBX can successfully send calls out.

Accept Push Notifications

By default, the user can receive push notifications on Linkus Mobile Client anywhere and anytime, such as missed calls, new voicemail messages and so on.

Setting	Description
Accept Push Notification	Enable or disable push notifications on Linkus Mobile Client.

Agent Status Auto Switch

If the user is a dynamic agent who needs to frequently log in to or out of a queue, you can associate queue status with extension presence. The user's status in a queue will automatically change along with his or her extension presence.

Setting	Description
Login	Log in to a queue.

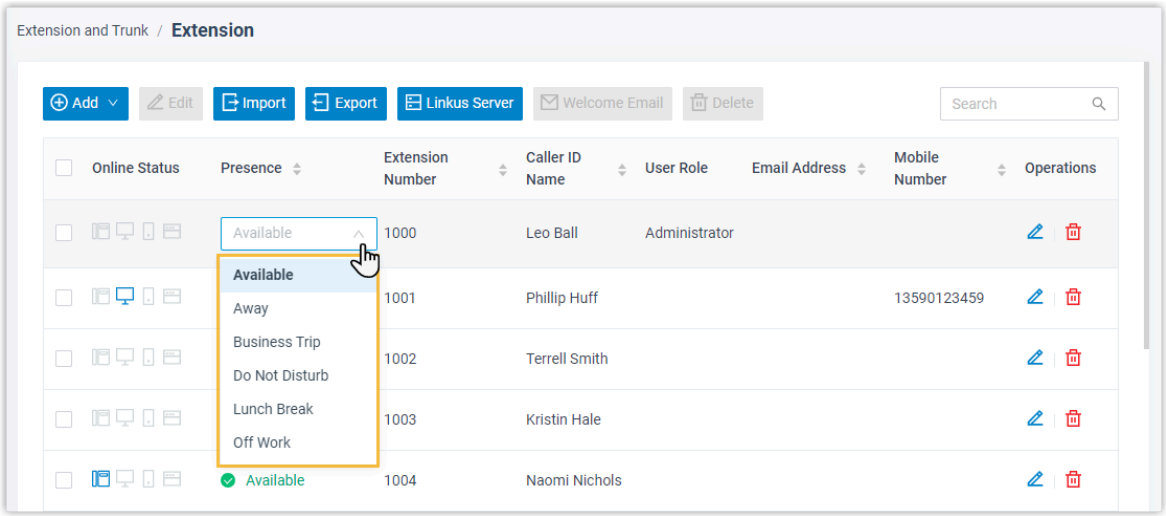
Setting	Description
	Note: The option is available ONLY in **Available** status.
Logout	Log out of a queue.
Pause	Pause receiving queue calls. **Note:** If you have set pause reasons for queue agents, you can select a specific pause reason in the **Pause Reason** drop-down list.
Do Nothing	Retain current status.

Manually Switch Extension Presence

This topic describes how to switch an extension's presence manually.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, find the desired extension.
3. In the **Presence** column, select a status from the drop-down list.



4. On the current page, click a blank space.
5. Click **Apply**.

Result

New presence is synchronized on Linkus clients; presence settings related with the status take effect.

Related information

Automatically Switch Extension Presence Based on Time

Automatically Switch Extension Presence Based on Time

This topic gives a configuration example to describe how to configure presence auto switch based on Business Hours and Holidays for specific extension users.

Background information

Assume that you have set Business Hours and Holidays on the PBX system, and you want the presence of extensions to be automatically switched according to the following time schedule:

Business Hours and Holidays	Time-based Presence
Business Hours: 09:00-12:00 and 14:00-18:00 from Monday to Friday.	Available
Break Hours: 12:00-14:00 from Monday to Friday.	Lunch Break
Holidays: December 25 to January 5.	Off Work
Outside Business Hours: The time periods that are not defined as Business Hours, Break Hours, or Holidays.	Off Work

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the extensions that need to switch presence status automatically based on the time schedule.
2. Click the **Features** tab, and go to **Time-conditional Presence Auto Switch** section.
3. Configure the following presence based on the time:
   - **Business Hours**: Select a status to be displayed during office hours.

     In this scenario, select **Available**.
   - **Break Hours**: Select a status to be displayed during break time.

     In this scenario, select **Lunch Break**.
   - **Holidays**: Select a status to be displayed during holiday.

     In this scenario, select **Off Work**.

- **Outside Business Hours**: Select a status to be displayed during non-office hours.

  In this scenario, select **Off Work**.

4. Click **Save**.

> **Note:**
> The priority of presence switching at different times is: **Holidays** > **Break Hours** > **Business Hours** > **Outside Business Hours**.

Result

Presence status will be switched automatically based on time.

For example, after 18:00, the presence displayed on Linkus client will be switched to **Off Work**.

> **Note:**
> If someone overrides time condition for the system, the presence status will be switched accordingly.
>
> For example, time condition is overridden to **Business Hours**, the presence status will be force switched to **Available**.

Related information

Overview of Business Hours and Holidays
Manually Switch Extension Presence

Monitor Extension Status by BLF Key

This topic describes how to configure a BLF key for auto-provisioned IP phone on PBX web portal, so as to monitor the call status and DND (Do Not Disturb) presence status of a specific extension.

Prerequisites

The phone is connected to Yeastar P-Series Cloud Edition via Auto Provisioning, and has been assigned an extension.

For more information, see Auto Provision IP Phones.

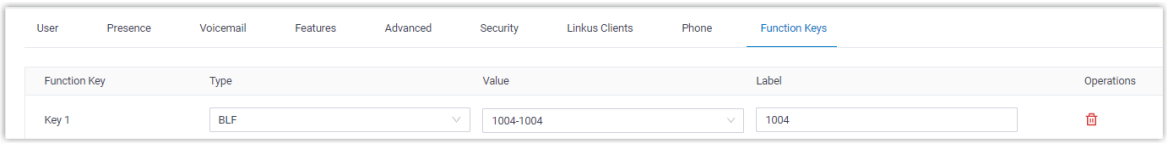
Procedure

- Step 1. Set up a function key for extension monitoring
- Step 2. Apply the configuration to the IP phone

Step 1. Set up a function key for extension monitoring

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the extension that is assigned to the phone.
2. Click the **Function Keys** tab.
3. Configure a function key to monitor the status of an extension.

   The following figure shows a configuration example of monitoring extension 1004.

   - **Type**: Select **BLF**.
   - **Value**: In the drop-down list, select an extension to monitor. In this example, select `1004`.
   - **Label**: Optional. Enter a value, which will be displayed on the phone screen.
4. Click **Save**.

Step 2. Apply the configuration to the IP phone

1. Go to **Auto Provisioning > Phones**, click beside the desired phone.

   The system prompts you whether to reprovision the phone.
2. In the pop-up window, click **OK**.

Result

- The LED of the BLF key shows the real-time status of extension 1004:
  - **Solid Green**: The extension is being monitored, and the status is idle.
  - **Solid Red**: The extension is sending a call or is in a call; or the extension presence is DND (Do Not Disturb).

> **Note:**

> For Fanvil IP phones that support differentiated DND status indication, the DND status is indicated by a **Solid Yellow** LED light. For more information regarding the supported phone models and firmware versions, contact your Fanvil IP phone provider.

  - **Flashing Red**: The extension is ringing.
  - **LED off**: The extension is not registered, or the extension has been deleted from the PBX system.

- You can press the BLF key on the phone to achieve the followings:
  - Place a call to the monitored extension.
  - Pick up the monitored extension's incoming calls.

> **Note:**
> To achieve this, make sure that the Extension Pickup feature code is enabled (Path: **Call Features > Feature Code > Call Pickup > Extension Pickup**).

Related information

Pick up a Call for a Group Member
Pick up a Call for a Specific Extension
Linkus Web Client Guide - Configure Function Keys
Linkus Desktop Client Guide - Configure Function Keys
IP Phone Configuration Guide - Monitor Extension Status by BLF Key on Fanvil IP Phone

Forward Incoming Calls to Another Destination

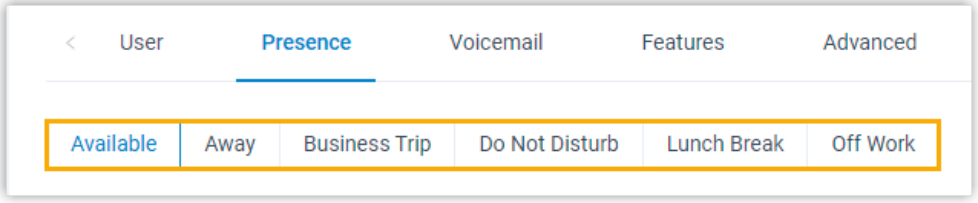
Yeastar P-Series Cloud Edition supports status-based call forwarding, which allows users to forward incoming calls to different destinations based on their presence status. This topic describes how to preconfigure call forwarding rules for extension users on PBX, and how to enable and configure feature code so that extension users can make immediate changes to call forwarding destinations by dialing a feature code when needed.

Set up call forwarding (destination preset)

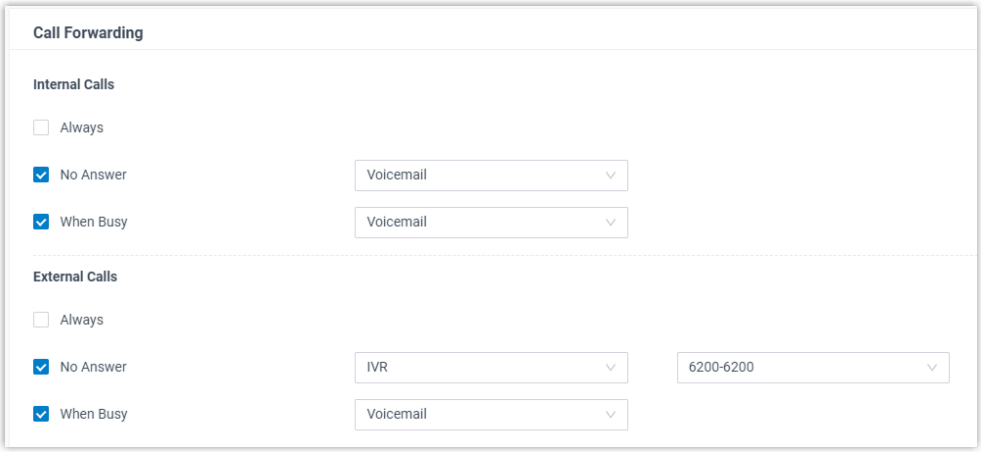
For each presence status of an extension, you can define a different destination to which the incoming calls will be forwarded. Every time the presence status changes, the incoming calls will be forwarded to the corresponding destination.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Presence** tab.
3. In the status bar, select a status to which the call forwarding rule will be applied.



4. In the **Internal Calls** and **External Calls** section, select a forwarding action and specify a destination.



5. Click **Save** and **Apply**.

Result

When a call reaches the extension number, the system will check the user's presence, identify whether it originates from an internal caller or external caller, and then route the call to the specified destination.

Enable feature code for call forwarding (destination immediate change)

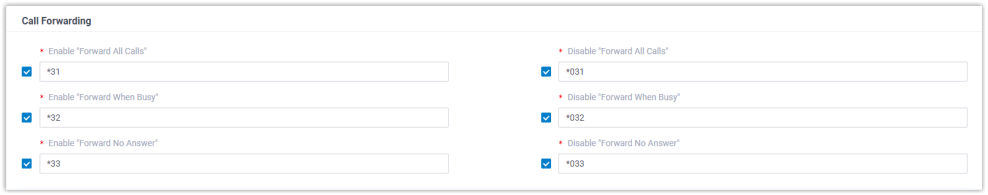
Extension users can change the preset call forwarding destinations on Linkus Web Client. For users with no access to Linkus Web Client, you can enable feature codes for call forwarding, so that these users can dial a feature code on their phones to change the call forwarding destinations.

Restrictions

- Be it internal calls or external calls, all the incoming calls received under the same forwarding type (**Always**, **No Answer**, and **When Busy**) will be routed to the same destination.
- Extension users can only change the call forwarding destination of their current presence status.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. In the **Call Forwarding** section, enable and configure the feature codes for call forwarding as needed.



   - **Enable ''Forward All Calls''/''Forward When Busy''/''Forward No Answer''**: Select the checkbox, then configure a feature code.

     Extension users can dial the feature code to forward calls to voicemail or a specific number. For more information, see Call Forwarding Feature Code.
   - **Disable ''Forward All Calls''/''Forward When Busy''/''Forward No Answer''**: Select the checkbox, then configure a feature code.

     Extension users can dial the feature code to disable automatic call forwarding.
3. Click **Save** and **Apply**.

Result

When an extension user dials the feature code, incoming calls received under the current presence status will be routed to the specified destination.

Note:

> Forwarding type **No Answer** and **When Busy** are not supported under **Do Not Disturb** presence status, which means that even if extension users dial the corresponding feature code, the configuration will not work.

Ring Office Phone and Mobile Phone Simultaneously

This topic describes how to achieve simultaneous ring on office phone and mobile phone.

Scenario

A user may miss important calls when he or she is away from desk or on a business trip. In this case, you can enable simultaneous ring for the user. When a call reaches the user's extension number, both mobile phone and office phone with the extension number logged in will simultaneously ring.

Prerequisites

- You have set a mobile number for the extension.
- At least one outbound route is ready for use.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Presence** tab.
3. In the status bar, select a status to which the strategy of simultaneous ring will be applied.
4. In the **Options** section, configure the following settings.
    a. Select the checkbox of **Ring the Mobile Number Simultaneously**.
    b. Click to configure mobile number.
    c. **Optional:** In the **Prefix** field, enter the prefix of outbound route so that PBX can successfully send calls to your phone.
        - If the **Strip** of outbound route is not set, you don't have to set the **Prefix**.
        - If the **Strip** of outbound route is set, you need to set the **Prefix** according to the **Patterns** of outbound route.
5. Click **Save** and **Apply**.

Result

If a call reaches the user's extension number when he or she is in the specified presence, both office phones and mobile phone will ring simultaneously.

Extension Voicemail

Set up Extension Voicemail

This topic introduces voicemail feature and describes how to set up voicemail for an extension.

Background information

Yeastar P-Series Cloud Edition supports voicemail feature, which helps users receive audio messages when they are unavailable to answer calls. When you create an extension, the voicemail feature is enabled by default, and a 4-digit PIN code is randomly generated for accessing voicemail.

You can retain default settings, or change the following settings according to your needs.

- Enable or disable voicemail feature
- Voicemail PIN Authentication
- Voicemail language
- Email notifications
- Play options of voicemails
- Voicemail greetings

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click  beside the desired extension.
2. Click **Voicemail** tab, turn on the option **Enable Voicemail**.
3. In the **Voicemail PIN Authentication** drop-down list, decide whether a PIN is required when the user accesses voicemail.
    - **Enabled**: This user needs to enter a PIN to access his or her voicemail messages.

      You can set the PIN number in the **Voicemail Access PIN** field.

- **Disabled**: This user can access his or her voicemail messages directly.

4. In the **Voicemail Language** drop-down list, decide the system prompt language used by the extension's voicemail.

   When callers are routed to the extension's voicemail, a system prompt "The extension `{extension_number}` you dialed cannot be connected now. Please leave your message after the tone. When done, hang up, or press the pound key" will be played in the selected language.

   > **Note:**
   > - The available language is synchronized from System Prompt (Path: **PBX Settings > Voice Prompt > System Prompt**).
   > - If the extension's voicemail has already set up a greeting, the custom greeting will be played to callers instead.

5. In the **New Voicemail Notification** drop-down list, decide whether to send email notifications when the user receives a new voicemail.
   - To disable email notifications, select **Do Not Send Email Notifications**.
   - To enable email notifications, select one of the following options:
     - **Send Email Notifications with Attachment**: Send a notification email with the new voicemail message attached as a `.wav` file.
     - **Send Email Notifications without Attachment**: Send notification emails only.
6. If you enabled email notifications, configure the following settings as needed:

Setting	Description
After Notification	Decide how to deal with voicemails after notification emails are sent out. • **Mark as Read**: Mark the voicemail message in mailbox as read. • **Delete Voicemail**: Delete the voicemail messages from mailbox. • **Do Nothing**: Keep the voicemail message in mailbox as unread.
Send to	Specify the email address for receiving notification emails. • **User Email**: Send notification emails to the user's email address. • **Custom Email**: Send notification emails to a custom email address.

Setting	Description
	Enter the desired email address in the **Custom Email Address** field.

7. **Optional:** Set whether to play the following messages when playing a voicemail.
   - **Play Date and Time**
   - **Play Caller ID**
   - **Play Message Duration**
8. **Optional:** To customize voicemail greetings that will be played to callers when they reach the user's voice mailbox, see Record or Upload Voicemail Greetings.
9. Click **Save** and **Apply**.

Related information

Forward Voicemail Messages to Email
Check Voicemail Messages

Extension Features

Handle Incoming Calls Based on Caller ID

This topic describes how to create a call handling rule for a specific user to handle incoming calls (calls from colleagues and external contacts) based on incoming Caller ID.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Features** tab.
3. In the **Call Handling Based on Caller ID** section, set up one or more rules according to your needs.
   a. Click **Add**.
   b. In the **Caller ID** field, enter a specific number or a number pattern.
      - To apply the rule to a specific number, enter a specific number.

        For example, enter `10086` to handle incoming calls with Caller ID 10086 based on the rule.
      - To apply the rule to a number pattern, enter a wildcard pattern.

For example, enter `9011.` to handle incoming calls with any Caller ID starting with 9011 based on the rule.

For more information, see Caller ID Pattern.

c. In the **Action** drop-down list, set how to deal with incoming calls with the Caller ID.

- **Hang Up**
- **Extension**
- **Voicemail**
- **IVR**
- **Play Greeting then Hang up**
- **Accept Call**

> **Note:**
> By default, all incoming calls are allowed to reach the extension. If there is a call-handling rule to prevent spam calls (eg.728373XX) from reaching the extension, but the extension user wants to accept calls from a specific number (eg.72837300), you can create another rule to accept calls from 72837300.

d. Click **Save**.

e. **Optional:** To add more rules, repeat **step a-d**.

f. **Optional:** In the **Move** column, adjust the rules' order. The rules take effect from the top down.

> **Note:**
> For example, set the rule "Accept calls from 72837300" to a higher priority than the rule "Reject calls from numbers starting with 728373".
> In this way, when receiving calls from 72837300, the system will send calls to the extension user. For other incoming calls from number starting with 728373, the system will hang up directly.

4. Click **Save** and **Apply**.

Result

When incoming calls reach the extension, the system will handle the calls based on Caller IDs.

Set up Email Notifications for Missed Calls

To remind an extension user of missed calls, you can set up email notifications of missed calls for the extension user.

Prerequisites

- System email server is set up.
- An email address is associated with a desired extension.

Procedure

1. Log in to PBX management portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Features** tab.
3. In the **Notifications** section, select the checkbox of **Send email notifications on missed calls**.
4. Click **Save** and **Apply**.

Result

If the extension user has missed calls, system will send notification emails to the user's mailbox.

> **Note:**
> If the extension user is a ring group member, and Record Missed Calls feature is enabled for the ring group, they system will also send notification emails when the user has missed calls from the ring group.

Set up Email Notifications for User Password Change

To remind an extension user of user password change, you can set up email notifications of user password change for the extension user.

Prerequisites

- System email server is set up.
- An email address is associated with a desired extension.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Features** tab.
3. In the **Notifications** section, select the checkbox of **Send email notification when the User Password is changed**.
4. Click **Save** and **Apply**.

Result

If the extension user's user password has been changed, system will send notification emails to the user's mailbox.

Customize Music on Hold for an Extension

This topic describes how to customize music on hold for a specific extension on Yeastar P-Series Cloud Edition, including the music played when an extension user holds a call and when a call is forwarded to another destination.

By default, extensions' music on hold follows the system's global prompt settings (set on **PBX Settings > Voice Prompt > Prompt Preferences > Music on Hold / Music on Hold for Call Forwarding**). You can refer to this topic to customize the prompts for individual extensions to provide a more personalized experience.

Requirements

The firmware version of PBX server is 84.15.0.22 or later.

Prerequisites

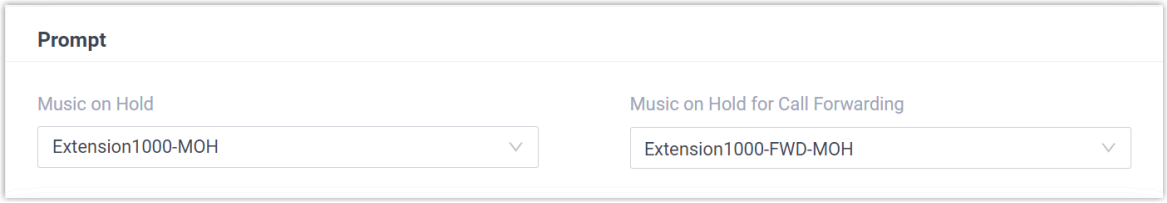
You have prepared and uploaded music for the following scenarios:

- **Music on Hold**: The music played to the other party when the extension user holds a call.
- **Music on Hold for Call Forwarding**: The music played to the caller when the call is forwarded to another destination.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.

2. Click beside the desired extension, then go to **Features > Prompt**.
3. In the drop-down list of the music on hold settings, select the desired music that you have uploaded.



4. Click **Save** and **Apply**.

Allow Multiple Registrations for One Extension Number

Registering one extension number to multiple SIP endpoints allows the employees to handle calls on any devices. This topic describes how to set the maximum concurrent registrations for an extension.

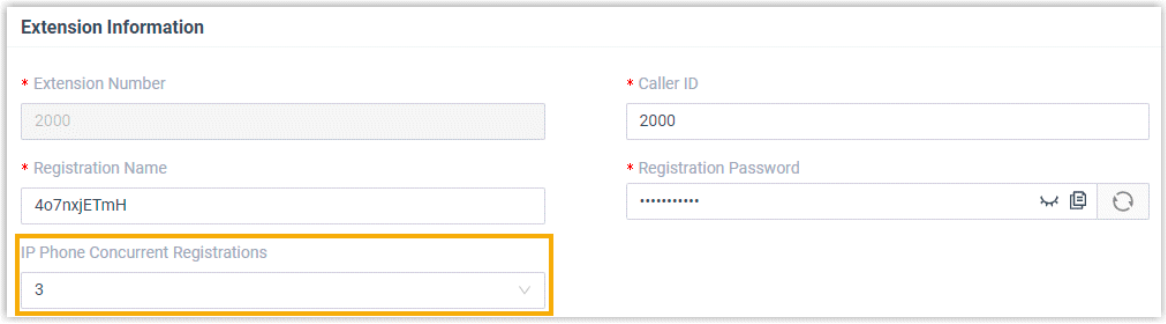
Background information

For employees who work flexibly anywhere, they can register their extensions on multiple SIP endpoints, such as an IP phone in their office, a softphone on computer, or a SIP client on mobile phone. In this way, an incoming call can ring all endpoints at the same time, and users can handle calls at anywhere on any devices.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. In the **Extension Information** section, set the maximum endpoints allowed to register the extension in the **IP Phone Concurrent Registrations** field.
   In this example, set the concurrent registrations to 3.

> **Note:**
> - The maximum number of concurrent registration is 3.
> - Concurrent Registration setting only limits the registration number of non-Linkus SIP endpoints. The registration number of Linkus clients is not counted.



3. Click **Save** and **Apply**.

Result

In addition to being registered on Linkus clients, the extension can also be registered on 3 other SIP endpoints.

When the extension receives a call, all the endpoints will ring. The extension user can handle the calls on any endpoint.

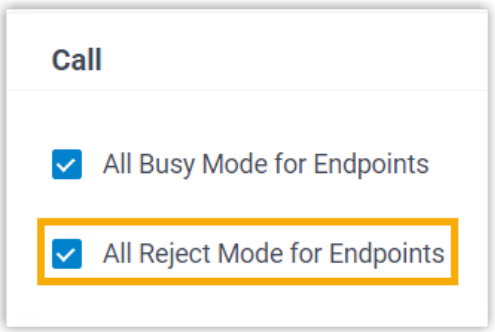
> **Note:**
> By default, when the extension is busy in a call and a new call reaches, all the endpoints (Linkus and other SIP endpoints) can still ring.
>
> To prevent other endpoints from receiving a new incoming call when an endpoint is busy, go to **Extension and Trunk > Extension > Features > Call** to enable **All Busy Mode for Endpoints** for the extension.

Stop Rejected Calls from Ringing Other Endpoints

If a user's extension has been registered on multiple endpoints, when the user rejects an incoming call on one of the endpoints, the call keeps ringing all the other endpoints. In this case, you can set up the extension to stop rejected calls from ringing other endpoints.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click the **Features** tab.
3. In the **Call** section, select the checkbox of **All Reject Mode for Endpoints**.



4. Click **Save**.

Result

When the extension user rejects an incoming call on an endpoint, the other endpoints will stop ringing. The call will be routed to the extension's **When Busy** destination (Path: **Extension and Trunk > Extension > Presence > Call Forwarding**).

Set up Third-party Integration for Call Popup

Yeastar Popup URL allows a lightweight integration with a third-party application (such as CRM system, ERP system, etc.) to achieve call popup. When an extension receives a call, the PBX calls the URL of the third-party application and retrieves relevant customer data to display on the pop-up web page.

Restrictions and requirements

Restrictions

The feature only works when Linkus Web Client is logged in.

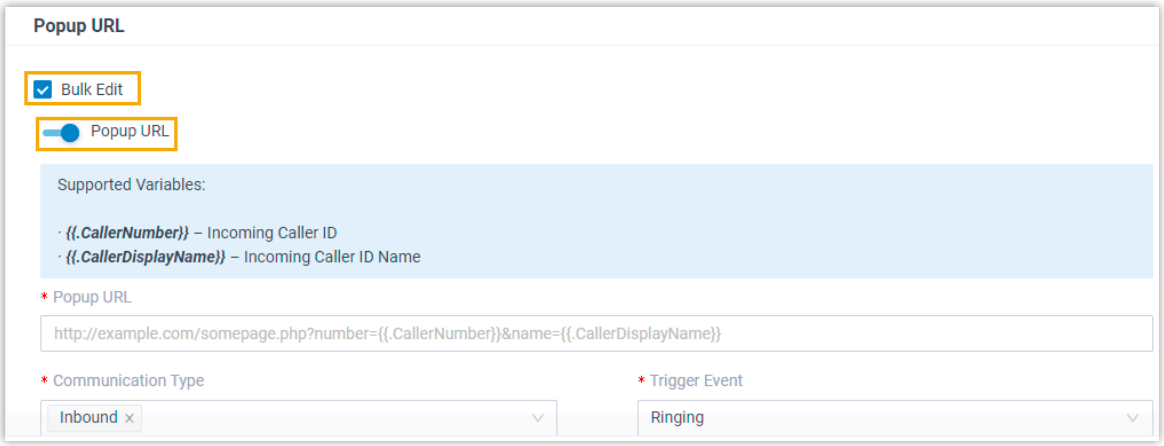
Requirements

- **Third-party application**:
  - Web-based.
  - Support to provide a URL that can identify callers via Caller ID and Caller ID Name.

Procedure

Follow the instructions below to set up popup URL for extensions in bulk. You can also customize it for a specific extension.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Select the checkboxes of the desired extensions, click **Edit**.
3. Under the **Features** tab, select the checkbox of **Bulk Edit** and turn on the option **Popup URL**.



4. Set up the third-party integration via Popup URL.

Table 4.

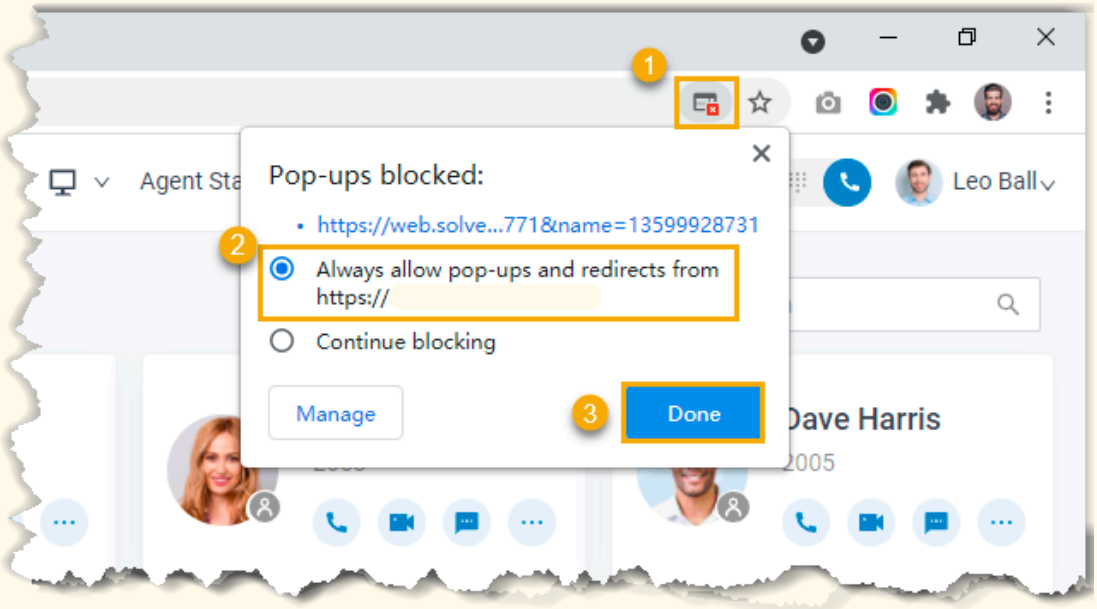
Settings	Descriptions
Popup URL	Enter the third-party URL, followed by the variables that you want to pass. **Supported variables**: • `.{{.CallerNumber}}`: Incoming Caller ID. • `.{{.CallerDisplayName}}`: Incoming Caller ID Name. Take Solve360 CRM as an example: https://web/solve.360.com/{{.CallerNumber}}&{{.CallerDisplayName}}
Communication Type	Select which types of calls will trigger the call popup. • **Inbound**: Inbound calls from external users. • **Internal**: Internal calls from colleagues.
Trigger Event	Set when the call popup will be automatically triggered. • **Ringing**: An incoming call reaches. • **Answered**: An incoming call is answered. • **Call End**: An incoming call is ended.

5. Click **Save**.

Result

When an incoming call reaches the extensions on Linkus Web Client, a pop-up screen automatically appears in the web browser and displays relevant customer data.

Important:
For the first-time use, users need to allow pop-ups and redirection from Linkus Web Client, or the pop-up screen can NOT be opened automatically.



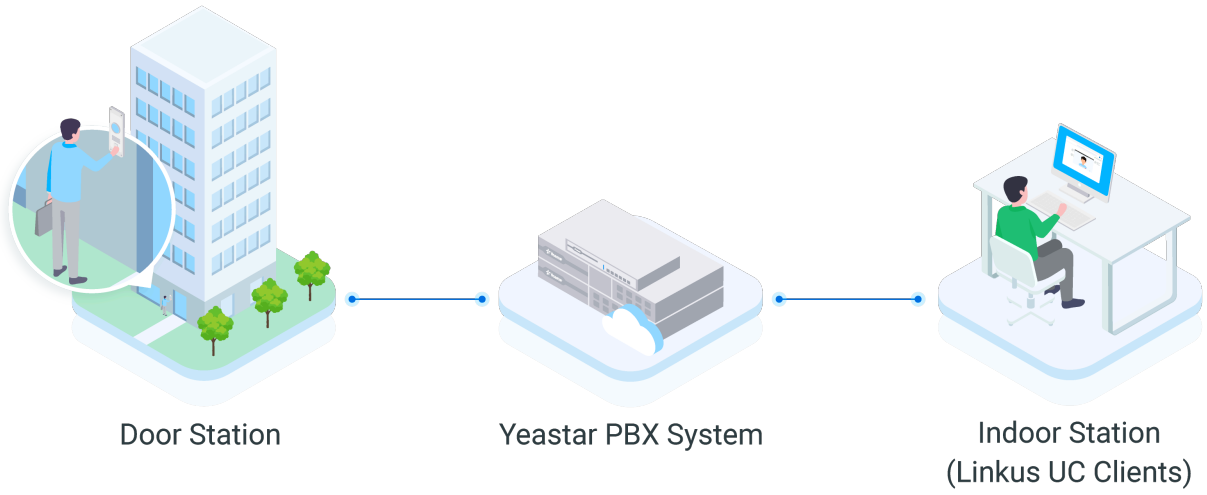
Enable Video Preview of Intercom Calls for Extensions

If you integrate video intercom system with Yeastar P-Series Cloud Edition, you can enable video preview of intercom calls for the extension registered on the door station. In this way, when receiving an incoming call from the door station, users who have their extensions registered as the indoor station can preview the visitor's video on their Linkus UC clients while ringing, and control the opening of the door.

Scenario

A video intercom system contains a door station and an indoor station. If you integrate video intercom system with Yeastar P-Series Cloud Edition, you can make the door station a regular PBX extension for making calls by registering an extension on it, and Linkus UC clients can work as the indoor station.

When visitors press the door station to ring the bell, the door station sends the doorbell (namely a call) to the indoor station. Users can preview the video on Linkus client while ringing, and decide whether to open the door for visitors.



Methods of previewing videos

Yeastar P-Series Cloud Edition supports the following two methods of previewing videos:

- **Video Preview**: This method requires action from callee. When receiving an incoming call from door station (with video preview enabled on the extension), the callee can click the preview button on the Linkus client to preview the video while ringing.
- **Auto Preview**: This method does NOT require action from callee. When receiving an incoming call from door station (with auto preview enabled on the extension), the callee's Linkus client will automatically show the video while ringing.

Requirements and restrictions

Before you begin, read through the requirements and restrictions for the feature:

Requirements

- **PBX Version**: 84.10.0.30 or later
- **PBX Plan**: Ultimate Plan

Restrictions

- **Supported Door Station**: Fanvil

Note:

> Fanvil door stations have been tested and proven to interoperate with Yeastar P-Series Cloud Edition. For other door stations, please contact Yeastar.

- **Supported Indoor Station**: Linkus UC Clients

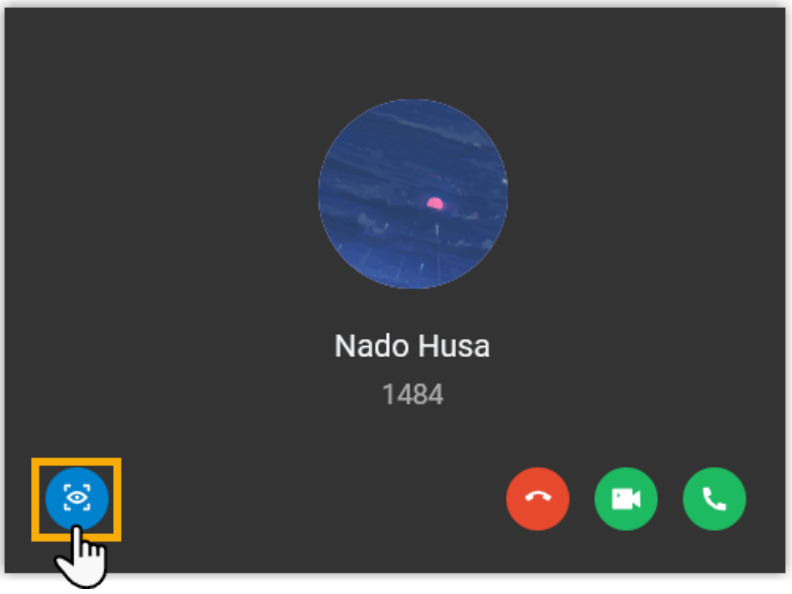
  Take note that the supported video preview methods vary depending on Linkus clients. For more information, see the following table:

	Video Preview	Auto Preview
Linkus Mobile Client (iOS)	×	×
Linkus Mobile Client (Android)	√	√
Linkus Desktop Client (Windows)	√	× (Coming Soon)
Linkus Desktop Client (Mac)	√	× (Coming Soon)
Linkus Web Client	√	√

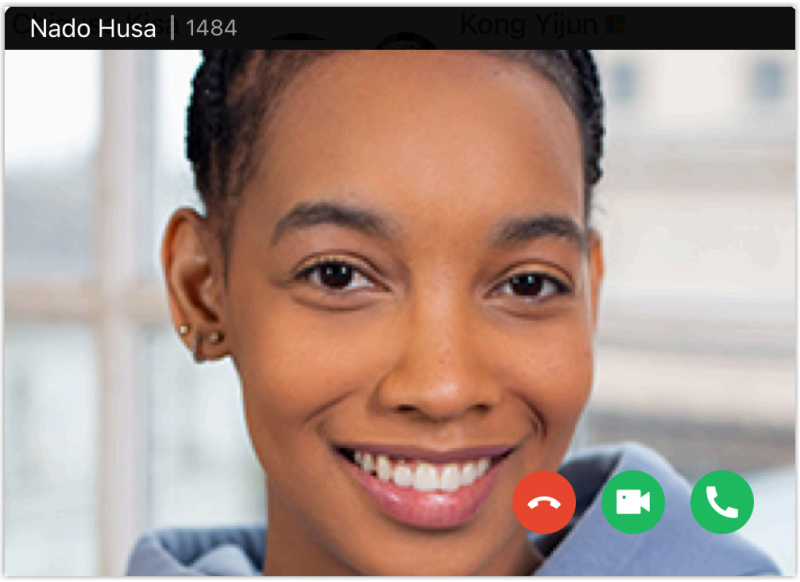
> **Note:**
> - If **Linkus Web Client** is set up to work with 'Yeastar Linkus for Google', the version of the Chrome extension must be **4.2.1** or later.
> - The version of **Linkus Android Client** must be **5.3.12** or later.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the extension for door station.
2. Click **Features** tab.
3. In the **Door Phone Application** section, set up the video preview feature:
   - **Video Preview**: If enabled, when receiving an incoming call from door station with the extension registered, the callee can click  on his or her Linkus client to preview the video while ringing.



- **Auto Preview**: If enabled, when receiving an incoming call from door station with the extension registered, the callee's Linkus client will automatically show the video while ringing.



4. Click **Save** and **Apply**.

Extension Advanced Settings

Advanced Settings of SIP Extension

This topic describes the advanced settings of a SIP extension.

> **Note:**
> The SIP configurations require professional knowledge of SIP protocol. Incorrect configurations may cause calling issues on the SIP extension.

Table 5.

Setting	Description
DTMF Mode	Set the mode for sending DTMF tones. • **RFC4733 (RFC2833)**: DTMF will be carried in the RTP stream in different RTP packets. • **Info**: DTMF will be carried in the SIP info messages. • **Inband**: DTMF will be carried in the audio signal. • **Auto**: If the device supports **RFC4733 (RFC2833)**, PBX will choose **RFC4733 (RFC2833)**, otherwise the PBX will choose **Inband**.
Transport	Set the transport protocol. • **UDP** • **TCP** • **TLS** **Note:** If you change the transport protocol, you must re-register the extension.
Qualify	Enable this option to send SIP OPTION packet to SIP device to check if the device is up.
T.38 Support	Enable or disable T.38 fax for the extension. **Note:** Enabling T.38 will add performance cost. We recommend that you disable T.38.
SRTP	Enable SRTP for voice encryption.

Extension Security

Extension Security Overview

This topic describes security options to prevent Yeastar P-Series Cloud Edition from unauthorized SIP registrations and abused outbound calls.

SIP security options

Yeastar P-Series Cloud Edition provides the following options to prevent unauthorized SIP registrations.

SIP User Agent Identification

By default, PBX allows phones to register extensions without user agent limit. To enhance extension security, you can restrict which user agent is allowed to register an extension.

When a phone is trying to register the extension, the phone will send SIP packets containing user agent. If the prefix of the user agent does not match the specified value, the registration will fail.

SIP Registration IP Restriction

By default, PBX allows SIP registrations without the limit of IP address.

To enhance extension security, you can specify which IP address or IP section is allowed to register an extension.

Call restrictions options

Yeastar P-Series Cloud Edition provides the following options to prevent abused outbound calls.

Disable Outbound Calls

Restrict users from making outbound calls.

Disable Outbound Calls outside Business Hours

Restrict users from making outbound calls during off-duty time and holidays.

Disallow International Calls

Restrict users from making international calls.

Note:

> The option works only when you have enabled **Enable Allowed Country/Region Code Dialaing Protection**. For more information, see Block Outbound International Calls.

Outbound Call Frequency Restriction

When an extension makes outbound calls and the number of calls exceeds the outbound call frequency restriction within specified time period, the system would restrict the extension from making outbound calls. For more information, see Limit Outbound Call Frequency of an Extension.

Max Outbound Call Duration (s)

When the user is in an outbound call and the call duration reaches the limit, the system would end the call.

Outbound Route Permission

Specify the outbound routes that an extension is allowed to use.

> **Note:**
> If this extension belongs to an organization or an extension group that has permission to use a specific outbound route, then you can't change the extension's permission to the outbound route here.

Restrict Outbound Calls for an Extension

Toll fraud is a global problem in telecommunication industry. It happens when hackers access your PBX system and make expensive phone calls from existing accounts. To prevent toll fraud, you can restrict outbound calls for an extension.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit a desired extension.
2. Click **Security** tab.
3. In the **Call Restrictions** section, select the checkbox of **Disable Outbound Calls**.
4. Click **Save** and **Apply**.

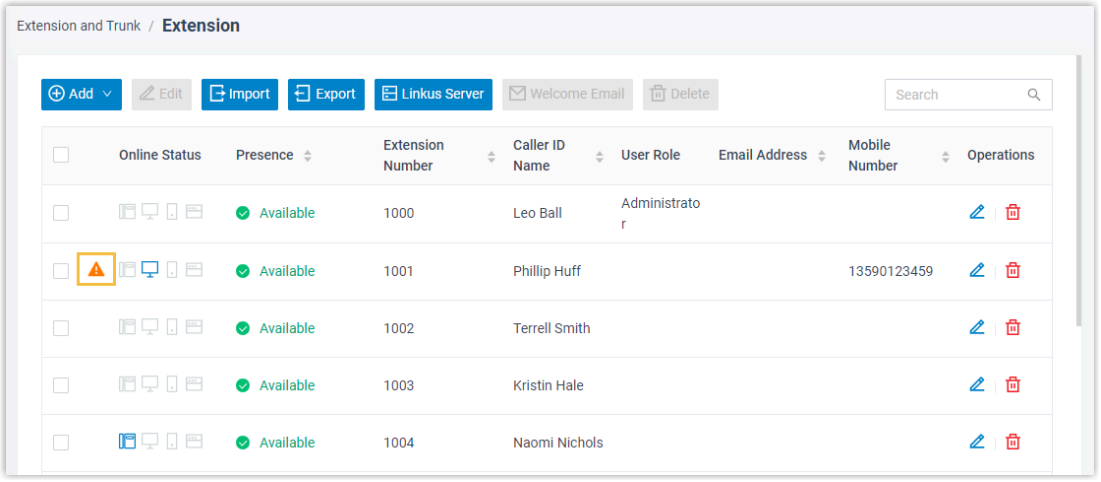
Result

- Users cannot make outbound calls even if the extensions are selected in outbound routes.

> **Note:**
> Emergency Calls like 911 is not restricted.

- On Extension list, ⚠ is displayed in front of the extension.

> **Note:**
>
> To cancel the restriction of outbound calls, click ✎ to edit the extension, go to **Security** tab and unselect the checkbox of **Disable Outbound Calls** in the **Call Restrictions** section.



Restrict Extension Registration Based on User Agent

This topic describes how to restrict extension registration based on user agent.

Background information

SIP is a peer-to-peer protocol. The peers in a session are called User Agents (UAs). A user agent can play one of the following roles:

- User Agent Client (UAC): A client application that initiates a SIP request, such as INVITE, ACK, OPTIONS, BYE, CANCEL, and REGISTER.
- User Agent Server (UAS): A server application that receives the SIP request from a UAC, and returns a response to the request back to the UAC.

When a SIP endpoint tries to register an extension to Yeastar P-Series Cloud Edition, the SIP endpoint working as UAC sends packets containing user agent string to the PBX. By default, Yeastar P-Series Cloud Edition allows registrations from any UAC without authenticating user agent. For security reasons, you can restrict extension registration based on user agent.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Security** tab.
3. In the **SIP Security** section, select the checkbox of **Enable User Agent Registration Authorization**.
4. Set the user agent.
    a. Click **Add User Agent**.
    b. In the **User Agent** field, enter a value.
5. Click **Save** and **Apply**.

Result

When a phone is trying to register an extension, the phone will send SIP packets containing a user agent, such as phone manufacturer, phone model, etc. If the prefix of the user agent does not match the specified value, the registration will fail.

Restrict Extension Registration Based on IP Address

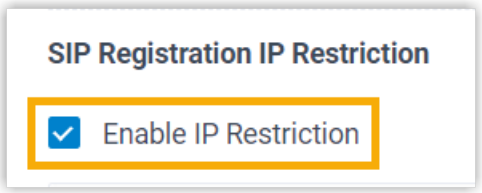
This topic describes how to allow devices with a specific IP address or in a specific IP section to register extensions on Yeastar P-Series Cloud Edition.

Background information

By default, Yeastar P-Series Cloud Edition allows SIP registrations without the limit of IP address. In case hackers remotely register extensions and make expensive phone calls, you can restrict that only devices with a specific IP address or in a specific IP section can register extensions on Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Security** tab.
3. In the **SIP Registration IP Restriction** section, select the checkbox of **Enable IP Restriction**.



4. Set which IP address or IP section is allowed to register the extension.
    a. Click **Add IP**.
    b. In the **Permitted IP** and **Subnet Mask** fields, set the allowed IP address or IP section.
5. Click **Save** and **Apply**.

Result

Only device with the IP address or in the IP section can register the extension.

Block Outbound Calls Outside Business Hours

This topic describes how to restrict an extension from making outbound calls outside business hours.

Prerequisites

You have set global business hours.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Security** tab.
3. In the **Call Restrictions** section, select the checkbox of **Disable Outbound Calls outside Business Hours**.
4. Click **Save** and **Apply**.

Result

The user can NOT make outbound calls during off-duty time and holidays.

Limit Call Duration of an Outbound Call

This topic describes how to limit call duration of an outbound call.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Security** tab.
3. In the **Call Restrictions** section, select a value from the drop-down list of **Max Outbound Call Duration (s)**, or enter a value according to your needs.
4. Click **Save** and **Apply**.

Result

When the user is in an outbound call and call duration reaches the **Max Outbound Call Duration (s)**, the system would end the call.

Limit Outbound Call Frequency of an Extension

To secure enterprise communications and reduce the economic loss if the PBX system has been hacked, we recommended that you set up rules to restrict the extension outbound call frequency.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Security** tab.
3. Scroll down to the **Call Restrictions** section, in the **Outbound Call Frequency Restriction** drop-down list, select the desired rule (s).

> **Note:**
> The PBX has a default rule **Default_Ext_Outbound Call Frequency**, which limits extension users to make maximum 5 outbound calls in 1 second. You

> can add new rules according to your need. For more information, see Add an 'Outbound Call Frequency Restriction' Rule.

4. Click **Save** and **Apply**.

Result

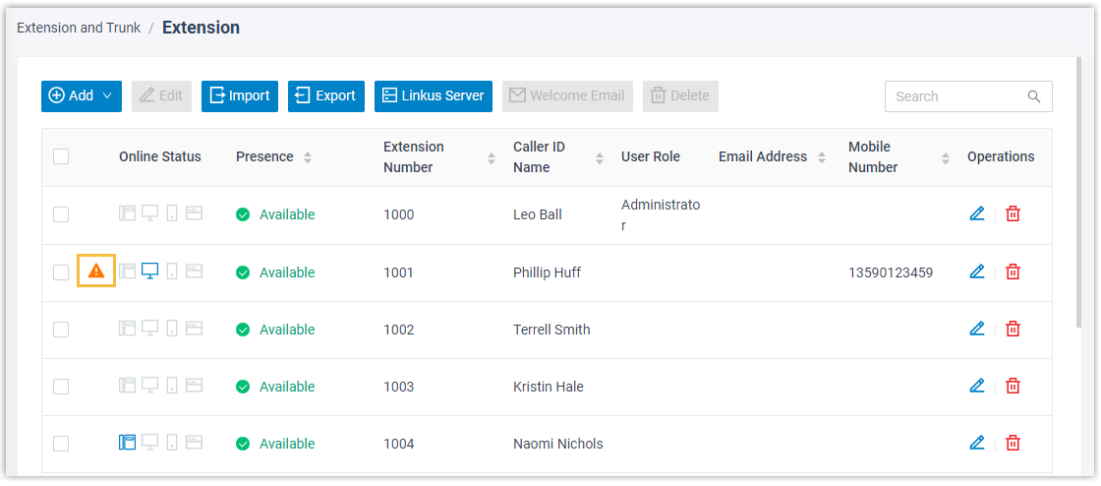
If an extension has exceeded the outbound call frequency restriction, the following things would happen.

- Users cannot make outbound calls even if the extensions are selected in outbound routes.

> **Note:**
> Emergency Calls like 911 is not restricted.

- On Extension list, ⚠ is displayed in front of the extension.

> **Note:**
>
> To cancel the restriction of outbound calls, click ✎ to edit the extension, go to **Security** tab and unselect the checkbox of **Disable Outbound Calls** in the **Call Restrictions** section.



- The system sends a notification to inform the notification contacts of an Outbound Call Frequency Exceeded event.

Manage Extensions

Edit Extensions

This topic describes how to edit an extension, or edit extensions in bulk.

Edit an extension

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, select the desired extension, click  .
3. Change extension settings according to your needs.
4. Click **Save** and **Apply**.

Bulk edit extensions

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, select the checkboxes of the desired extensions, click **Edit**.
3. Select the checkbox of the desired feature, change extension settings according to your needs.
4. Click **Save** and **Apply**.

Reset an Extension's User Password

An extension's user password is used to log in to PBX web portal and Linkus clients. As an administrator, you can reset an extension's user password if the user forgets password.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Search and select the desired extension, click  .
3. In the **User Information** section, delete the value in the **User Password** field, and enter a new password.
4. Click **Save**.

Result

The extension's user password is changed. You need to inform the user of the new password.

Export and Import SIP Extensions

The SIP extensions configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired extension information in the exported file, and import the file to PBX again. This topic describes how to export and import SIP extensions.

> **Note:**
> Only system super administrator can import SIP extensions.

Export all extensions

You can export all the SIP extensions to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Extension Parameters.

Import SIP extensions

We recommend that you export extension data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Extension Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, select your CSV file.
4. Click **Import**.

   The extension data in the CSV file will be displayed in the **Extension** list.

Related information

Import and Export -FAQ

Delete Extensions

This topic describes how to delete an extension or delete extensions in bulk.

Delete an extension

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, select the desired extension, click  .
3. In the pop-up dialog box, click **OK**.
4. Click **Apply**.

Bulk delete extensions

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, select the checkboxes of the desired extensions, click **Delete**.
3. In the pop-up dialog box, click **OK**.
4. Click **Apply**.

Extension Visibility Permission

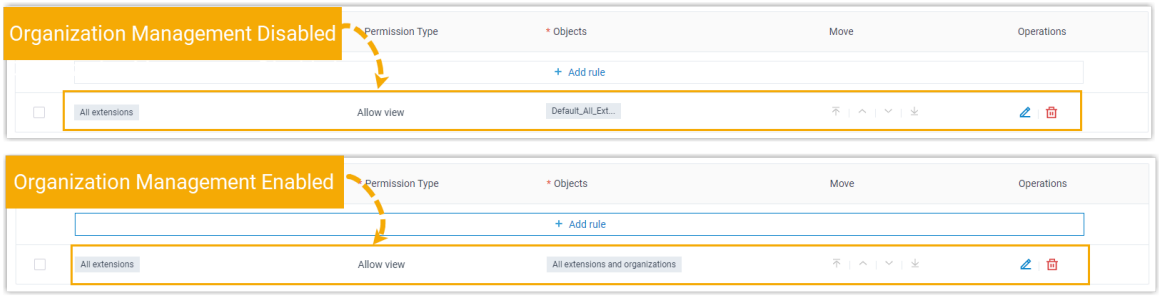
Set up Extension Visibility

By default, all the users can view all departments or the default extension group on Linkus clients, depending on whether you have enabled the organization management feature. To restrict users from viewing specific extensions, departments, or extension groups, you can set up extension visibility as the instructions provided in this topic.

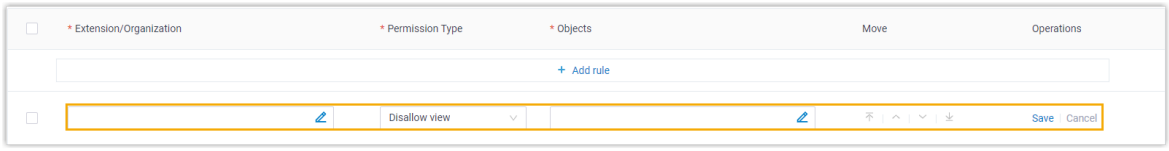
Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Extension Visibility**.

   The default extension visibility rule is displayed.



2. Click **Add rule** to create an extension visibility rule.
3. Set up the rule:



   a. Select desired values from the drop-down lists.

   - **Extension/Extension Group/Organization**: Click to select desired extensions, extension groups, or departments, for which you want to grant or restrict the viewing permission.
   - **Permission Type**: Select an option from the drop-down list to define the permission.
     - **Allow view**: Allow to view the extensions, extension groups, or departments, which are selected in the **Objects**.
     - **Disallow view**: Disallow to view the extensions, extension groups, or departments, which are selected in the **Objects**.
   - **Objects**: Click to select desired extensions, extension groups, or departments, which are allowed or disallowed to be viewed.

   > **Note:**
   > By default, when you select a department, its associated sub-departments are selected. Be careful when selecting departments.

   b. Click **Save**.

4. **Optional:** To adjust the rules order, click , , , or .

> **Note:**

> The priority of extension visibility rules is from the top down. When encountering permission conflicts, the permission is subject to the visibility rule with the higher priority.

Result

On Linkus clients, users can view the extensions, extension groups, or departments that are visible to them.

> **Note:**
> Users can NOT make calls to the extensions that they can not view.

Manage Extension Visibility Rules

This topic describes how to edit and delete extension visibility rules.

Edit an extension visibility rule

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Extension Visibility**.
2. Click beside a desired extension visibility rule.
3. Edit the rule as needed.
4. Click **Save**.

Delete extension visibility rules

> **Note:**
> Be careful when deleting extension visibility rules. If you delete all the extension visibility rules, all the internal calls would fail, as users can NOT make calls to the extensions invisible to them.

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Extension Visibility**.
2. To delete an extension visibility rule, do as follows:
   a. Click beside a desired rule.
   b. In the pop-up window, click **OK**.
3. To bulk delete extension visibility rules, do as follows:

a. Select the checkboxes of desired rules, click **Delete**.
b. In the pop-up window, click **OK**.

Contacts

Contacts Overview

Yeastar Contacts feature allows users to store external contacts outside of your company on PBX, access and call those contacts on endpoints (IP phone and Linkus clients) where their extensions have registered. This topic describes terminologies, requirements, key features, and limits of Yeastar Contacts feature.

Terminologies

Before using Yeastar Contacts feature, familiarize yourself with the following terminologies:

Personal Contacts

Personal Contacts is exclusive to each extension user, which allows users to store a number of personal contacts, such as direct customers.

> **Note:**
> Each user's Personal Contacts is only visible to himself or herself.

Company Contacts

Company Contacts is shared among authorized users, which allows authorized users to store a number of company shared contacts, such as company's customers, resellers, and partners.

Phonebooks

Phonebooks is a value-added service for Company Contacts, which allows authorized users to group company contacts into phonebooks, and implement robust control over access to each phonebook.

Key features

Group company contacts into phonebooks

Group company contacts into organized phonebooks and access them securely with robust permission control.

Sync contacts across Linkus clients

Sync contacts across Linkus Web Client, Desktop Client, and Mobile Client, and allow you to make changes on either client.

Sync contacts from integrated CRMs

Sync contacts from Customer Relationship Management (CRM) systems that are integrated with Yeastar P-Series Cloud Edition.

Identify incoming calls

When receiving an incoming call from a personal contact or a company contact, callee's phone screen will display the contact name. By knowing who is calling, users can handle calls efficiently.

For more information, see Identify Callers from Contacts.

Restrictions

Maximum Number of Extensions (N)	N ≤50	50 < N ≤ 200	N > 200
Company contacts (total)	50,000	200,000	500,000
Company phonebooks	100	200	500
Personal contacts (per extension)	100	200	300

Related information

Manage Company Contacts
Manage Company Phonebooks
Export and Import Company Contacts
Identify Callers from Contacts
Set up Contact Visibility
Allow Users to Query Contacts on IP Phones

Manage Company Contacts

This topic describes how to add, edit, or delete company contacts on PBX management portal.

Operation permissions

The authorized users can view and manage company contacts on Linkus clients, or view company contacts on an IP phone.

To grant users Company Contacts permissions, see Set up Contact Visibility.

To manage contacts on Linkus clients or an IP phone, see the following topics:

- Linkus Web Client - User Guide
- Use Contacts on an IP Phone

The following table shows what operations can be done on different endpoints.

Permission	Linkus Clients			IP Phone
	Web Client	Mobile Client	Desktop Client	
View company contacts	√	√	×	√
Add company contacts	√	√	×	×
Edit company contacts	√	√	×	×
Delete company contacts	√	√	×	×
Import company contacts	×	×	×	×
Export company contacts	×	×	×	×

Add a company contact

1. Log in to PBX web portal.
2. Go to **Contacts > Company Contacts**, click **Add**.
3. Enter contact information.
4. **Optional:** In the **Phonebook List** drop-down list, select one more phonebooks where you want the contact to be grouped.

   > **Note:**
   > - A newly created contact will be added to the default phonebook 'All Company Contacts_Phonebook', if any.

5. Click **Save**.

   The contact is stored in Company Contacts and synchronized to users' endpoints (IP phones and Linkus clients).

Edit a company contact

1. Log in to PBX web portal.

2. Go to **Contacts > Company Contacts**, click  beside the desired contact.
3. Edit contact information.
4. Click **Save**.

   Changes of the contact are synchronized to users' endpoints (IP phones and Linkus clients).

Delete company contacts

1. Log in to PBX web portal, go to **Contacts > Company Contacts**.
2. To delete a company contact, select the desired contact, click  and **OK**.
3. To delete company contacts in bulk, select the checkboxes of the desired contacts, click **Delete** and **OK**.

   The contacts are removed from Company Contacts and users' endpoints (IP phones and Linkus clients).

Manage Company Phonebooks

This topic describes how to add, edit, and delete company phonebooks.

Background information

Yeastar Phonebooks feature allows you to create phonebooks to group company contacts in an organized way and implement robust control over users' access to each phonebook.

Yeastar P-Series Cloud Edition supports two kinds of company phonebook:

- **PBX native company phonebook**: A phonebook that stores company contacts that are created on PBX web portal and Linkus Clients.
- **CRM-synchronized company phonebook**: A phonebook that stores company contacts that are synced from the integrated CRM. The phonebook will be marked with an identifier 'CRM' ().

> **Note:**
> Phonebooks synchronized from CRM can NOT be edited or deleted.

Group company contacts into phonebooks

1. Log in to PBX web portal.
2. Go to **Contacts > Phonebooks**, click **Add**.
3. In the **Phonebook Name** field, enter a name to help you identify it.
4. In the **Members** section, select desired company contacts.
   - **To define a All Contacts phonebook**:
     a. Select **All Company Contacts** from the drop-down list of **Select Contacts**.

     > **Note:**
     > Any time you add a company contact, the contact will be automatically added to the phonebook.

   - **To group contacts into a phonebook**:
     a. Select **Specific Company Contacts** from the drop-down list of **Select Contacts**.
     b. Click **Add** to select the desired company contacts.
     c. Click **Confirm**.
5. Click **Save**.

Edit company phonebooks

1. Log in to PBX web portal.
2. Go to **Contacts > Phonebooks**, click beside the desired phonebook.
3. Edit phonebook name, add or delete company contacts from the phonebook according to your needs.
4. Click **Save**.

   Changes of the phonebook are synchronized to users' Linkus clients.

Delete company phonebooks

1. Log in to PBX web portal, go to **Contacts > Phonebooks**.
2. To delete a phonebook, select the desired phonebook, click and **OK**.
3. To delete phonebooks in bulk, select the checkboxes of the desired phonebooks, click **Delete** and **OK**.
   The phonebooks are removed from PBX server and users' Linkus clients.

> **Note:**
> Company contacts in the phonebook are still kept in the system.

Related information

Contacts Overview
Set up Contact Visibility
Manage Company Contacts
Identify Callers from Contacts
Allow Users to Query Contacts on IP Phones

Export and Import Company Contacts

The company contacts configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired contacts in the exported file, and import the file to PBX again. This topic describes how to export and import company contacts.

Export company contacts

You can export all company contacts to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Contacts > Company Contacts**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Company Contacts Parameters.

Import company contacts

We recommend that you export company contacts data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 300 MB
- **File name**: Less than 127 characters

- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Company Contacts Parameters.

Procedure

1. Log in to PBX web portal, go to **Contacts > Company Contacts**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The company contacts in the CSV file will be displayed in the **Contacts** list.

Related information

Linkus Web Client Guide - Export personal contacts
Linkus Web Client Guide - Import personal contacts
Import and Export -FAQ

Identify Callers from Contacts

This topic describes how to configure Caller ID match to help users identify callers whose information is stored in Yeastar Contacts.

Background information

Caller ID match is supported on all kinds of endpoints, including Linkus clients, desk phones, or softphones. Yeastar P-Series Cloud Edition allows users to identify callers from Company Contacts and Personal Contacts.

Identify callers from Company Contacts

Support for authorized extension users who have permissions to view or manage company contacts.

For more information about how to grant permissions to users, see Set up Contact Visibility.

Identify callers from Personal Contacts

Support for each extension user.

Priority of Caller ID match

If an incoming number is stored in Company Contacts, Personal Contacts, mobile phone directory, and IP phone directory at the same time, the priority of Caller ID match from high to low is as follows:

- Mobile Phone Directory/IP Phone Directory
- Personal Contacts
- Company Contacts

Configure Caller ID match

1. Log in to PBX web portal, go to **Contacts > Company Contacts**.
2. Configure Caller ID match.
    a. On the **Company Contacts** page, click **Options**.
    b. Choose how to match incoming Caller ID.
        - **Do Not Match**: Display original incoming Caller ID.
        - **Exact Match**: Display contact name when an incoming Caller ID exactly matches existing number.
        - **Match the last *{number}* digits**: Display contact name based on the digits of incoming Caller ID.
            - If the digit length of an incoming Caller ID is shorter than or equal to the specified value, contact name will be displayed only when the incoming Caller ID exactly matches existing number.
            - If the digit length of an incoming Caller ID is longer than the specified value, contact name will be displayed when the last few digits of the incoming Caller ID matches that of existing number.

        > **Note:**
        > The default value is 7. To change the value, enter a number between 4 and 31.

    c. Click **Save**.

Caller ID match example

A contact Dora whose phone number is 12345678 is stored in Company Contacts; the system receives an incoming call from Dora.

- **Do Not Match** is selected:

    - When Dora calls in, the contact name "Dora" will not be displayed.
- **Exact Match** is selected:
    - If the incoming Caller ID is 12345678, the contact name "Dora" will be displayed.
    - If the incoming Caller ID is +012345678, the contact name "Dora" will NOT be displayed.
- **Match the last 9 digits** is configured:
    - If the incoming Caller ID is 12345678, the contact name "Dora" will be displayed.
    - If the incoming Caller ID is 15212345678, the contact name "Dora" will NOT be displayed.

Related information

Route Inbound Calls by Matched Phonebook Contacts

Allow Users to Query Contacts on IP Phones

To allow users to query contacts on IP phones, you need to auto provision IP phones. This topic describes how to allow users to query contacts on IP phones.

Requirements

IP Phone

Use Yealink phones of the required model and version. For more information, see Yealink phones.

> **Note:**
>
> - Yealink conference phones and DECT bases are NOT supported.
> - A maximum of 1000 company contacts and 300 personal contacts can be displayed on an Yealink phone.

Procedure

1. Grant permission for users to access company contacts.

   For more information, see Set up Contact Visibility.

> **Note:**
> By default, all the users have access to their own personal contacts, but no access to shared company contacts.

2. Synchronize contacts data to users' IP phones via Auto Provisioning.
   - If users' extensions haven't been associated with phones, see Auto Provision IP Phones to bind phones with the extensions.
   - If users' extensions have been associated with phones, reprovision the phones to take effect.
     a. Go to **Auto Provisioning > Phones**.
     b. Select the checkboxes of the desired phones, click **Reprovision**.

Result

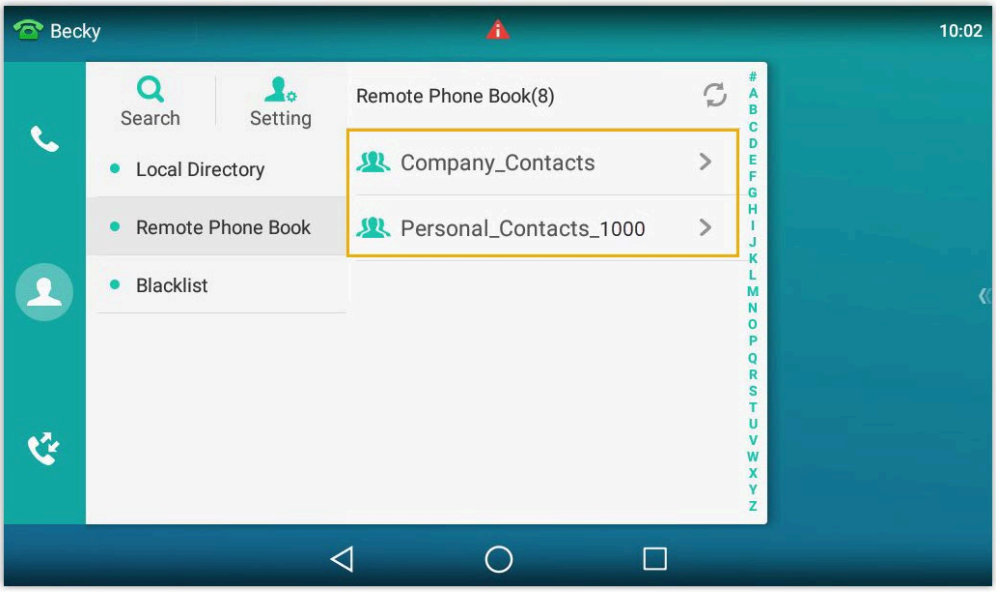
Contacts data are synchronized to IP phones' remote phonebooks. Users can query and place calls to contacts from the remote phonebook.

> **Note:**
> Two remote phonebooks from the PBX server are displayed on the IP phone:
>
> - **Company_Contacts**: Saves all the company shared contacts that you can view.
>
> > **Note:**
> > Company contacts on IP phones can NOT be grouped into phonebooks.
>
> - **Personal_Contacts_*{extension_number}***: Saves all your personal contacts.

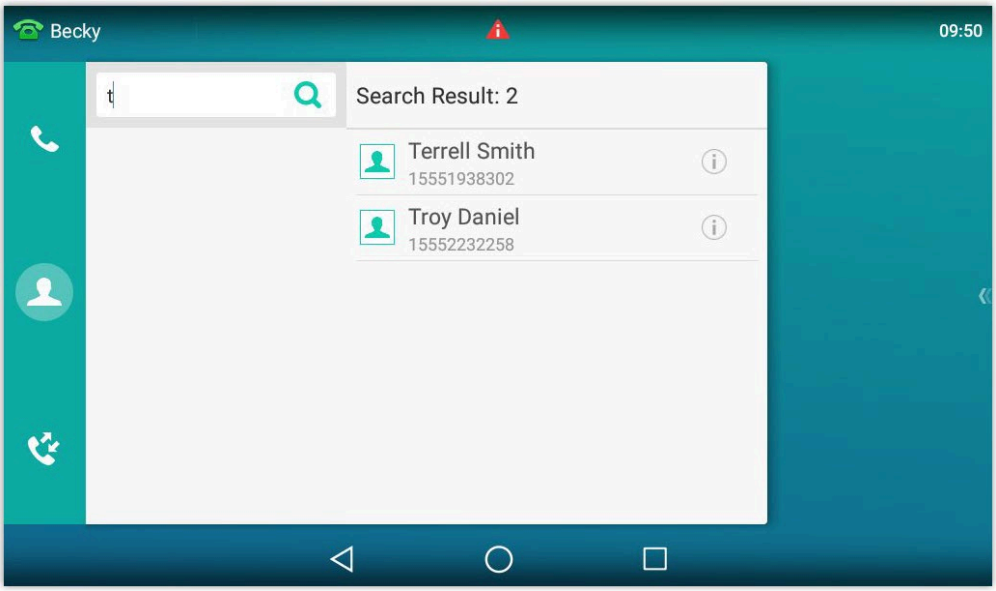
Example: Query contacts on Yealink T56A IP phone

1. Tap > **Remote Phonebook**.

   The directories that the user is allowed to view are displayed on the page.



2. Tap **Search**.
3. In the search box, enter contact name or number. The system will query contact from Contacts.



4. Select a contact, tap the contact number to quickly dial out.

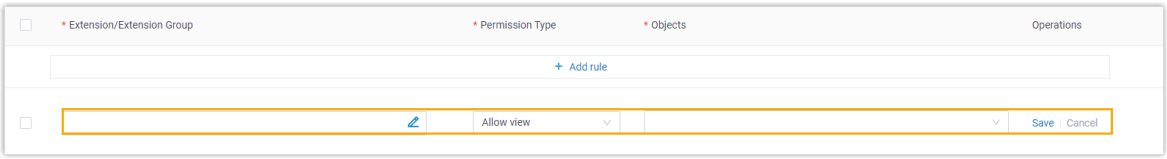
Contact Visibility Permission

Set up Contact Visibility

By default, all the users can neither manage nor view company contacts. To allow specific users to manage or view company contacts, you can set up company contacts visibility as the instructions provided in this topic.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Contact Visibility**.
2. Click **Add rule** to create a contact visibility rule.
3. Set up the rule:



   a. Select desired values from the drop-down lists.

   - **Extension/Extension Group/Organization**: Click to select desired extensions, extension groups, or departments, for which you want to grant the viewing permission or management permission.
   - **Permission Type**: Select an option from the drop-down list to define the permission.
     - **Allow view**: Allow to view the phonebooks that are selected in **Objects**.
     - **Allow manage**: Allow to view, add, edit, or delete the phonebooks that are selected in **Objects**.
   - **Objects**: Click to select desired phonebooks that are allowed to be viewed or managed.

   b. Click **Save**.

Result

- On Linkus clients, the authorized users can view or manage company contacts.

- On auto-provisioned Yealink IP phones, the authorized users can view company contacts.

  For more information, see Allow Users to Query Contacts on IP Phones.

Manage Contact Visibility Rules

This topic describes how to edit and delete contact visibility rules.

Edit a contact visibility rule

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Contact Visibility**.
2. Click  beside a desired contact visibility rule.
3. Edit the rule as needed.
4. Click **Save**.

Delete contact visibility rules

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Contact Visibility**.
2. To delete a contact visibility rule, do as follows:
   a. Click  beside a desired rule.
   b. In the pop-up window, click **OK**.
3. To bulk delete contact visibility rules, do as follows:
   a. Select the checkboxes of desired rules, click **Delete**.
   b. In the pop-up window, click **OK**.

Client Permission

Yeastar P-Series Cloud Edition Client Permissions

Client permissions enable you to control user access to specific information, features, and settings. Yeastar P-Series Cloud Edition supports customizing permission rules to control Linkus client menu and operation permissions, Extension visibility permission, and Contacts visibility permission.

Linkus clients menu and operation permissions

By default, users can access all the menus and configure all the settings within Linkus clients. You can set up permission rules to restrict users' access and operation permission:

- **Menu Visibility Permission**: Restrict users from specific menus within Linkus clients.
- **Operation Permission**: Restrict users from specific settings within Linkus clients.

For more information, see Set up User Permissions of Linkus Clients.

Extension visibility permission

By default, all the users can view all departments or the default extension group on Linkus clients. You can set up visibility rules to restrict users from viewing specific extensions, extension groups,or departments.

For more information, see Set up Extension Visibility.

Contact visibility permission

By default, all the users can neither manage nor view company contacts. You can set up visibility rules to allow specific users to manage or view company contacts.

> **Note:**
> Personal contacts are ONLY visible to the users themselves.

For more information, see Set up Contact Visibility.

LDAP Server

LDAP Server Overview

Yeastar P-Series Cloud Edition can be set as an LDAP Server, which provides centralized phonebook management. With this feature, you can store the contact information on the PBX, and quickly launch calls without wasting time finding a contact's number and subsequently entering it on your phone, thus greatly improving work efficiency.

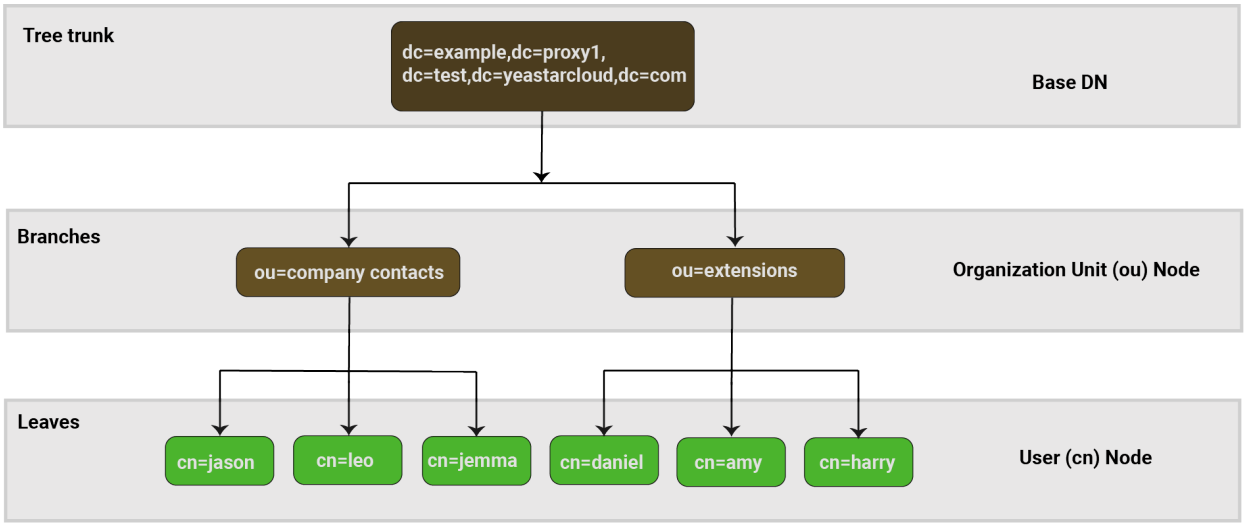
LDAP introduction

LDAP stands for Lightweight Directory Access Protocol, which is an application protocol for accessing and maintaining information services for the distributed directory over an IP network.

The LDAP directory server is based on the client/server mode. The LDAP Server contains directory data. An LDAP Client connects to the LDAP Server, and sends a request to obtain directory data from the LDAP Server, thus implementing global directory data management.

LDAP directory structure

The LDAP Server is a type of network database based on entries, which is a collection of information about an entity. In LDAP, directory entries are arranged in a hierarchical tree-like structure. The following figure shows an example of Yeastar P-Series Cloud Edition LDAP directory tree.



LDAP terminologies

An LDAP entry is a collection of information about an entity. Each entry consists of three primary components: a distinguished name, a collection of attributes, and a collection of object classes.

Distinguished Name (DN)

A globally-unique entry's distinguished name, which uniquely identifies the entry and its position in the directory information tree hierarchy.

A DN usually consists of three components.

- dc: Domain Component, usually refers to a component of the domain name.
- ou: Organization Unit, usually refers to a name of a group object.
- cn: Common Name, usually refers to a user name.

The DN of an LDAP entry is much like the path to a file on a filesystem. For example, `cn=amy,ou=extensions,dc=example,dc=proxy1,dc=test,dc=yeastarcloud,dc=com` is like a file path of `com/yeastarcloud/test/proxy1/example/extensions/amy`.

The **Base DN** is the root of the LDAP directory tree, which is the starting point of LDAP search. For example, `dc=example,dc=proxy1,dc=test,dc=yeastarcloud,dc=com`.

Attributes

Each entry can have multiple attributes. Each attribute has an attribute type and a set of values that comprise the actual data.

The syntax of values depends on the attribute type. The following table gives examples of attributes when `ou=company contacs`.

Table 6.

Attribute	Information details	Example
cn	Contact ID	Leo
displayName	Display Name	Leo Ball
givenName	First Name	Leo
sn	Last Name	Ball
mail	Email Address	leoball@example.com
company	Company	Yeastar

Table 6. (continued)

Attribute	Information details	Example
title	Job Title	Manager
department	Organization	Sales
telephoneNumber	Business number	+86-592-5503301
mobile	Mobile Number	12345678902
homePhone	Home Number	12345678902
facsimileTelephoneNumber	Fax Number	+86-592-5503301
postalCode	Zip Code	361024
l	City	Xiamen
st	Street	Software Park Phase III
co	Country	China

Object Classes

Object Class defines collections of attribute types which may be used in entries containing that class, and which of those attribute types will be required rather than optional. Every entry has a structural object class, which indicates what kind of object an entry represents (e.g., whether it is information about a person, a group, a device, a service, etc.), and may also have zero or more auxiliary object classes that suggest additional characteristics for that entry.

For example, if the objectclass is `person`, then the required attributes are `givenName` and `sn`, the optional attributes are `description`, `seeAlso`, etc.

Related information

Set up Yeastar P-Series Cloud Edition as an LDAP Server

Set up Yeastar P-Series Cloud Edition as an LDAP Server

This topic describes how to set up Yeastar P-Series Cloud Edition as an LDAP Server. In this way, you can store the contacts information on PBX and query from IP phones directly.

Procedure

1. Log in to PBX web portal, go to **Contacts > LDAP Server**.
2. On the top of the page, turn on **LDAP Server**.
3. Click the **LDAP Server Settings** tab to check the LDAP Server settings .

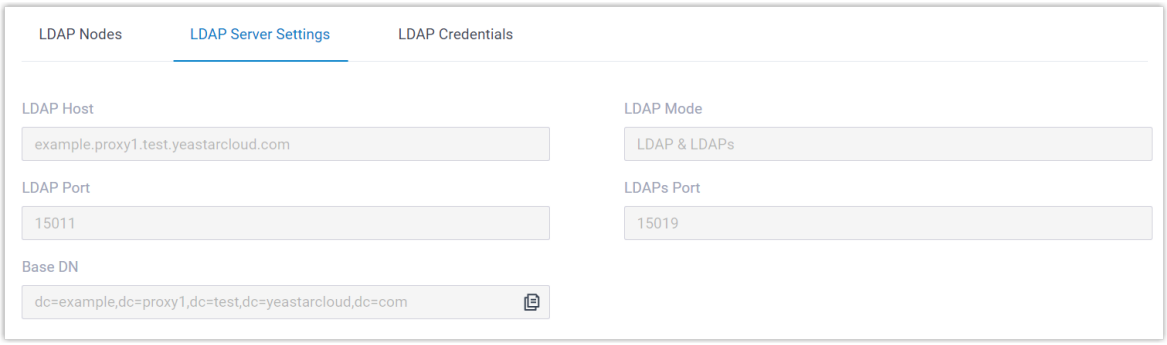
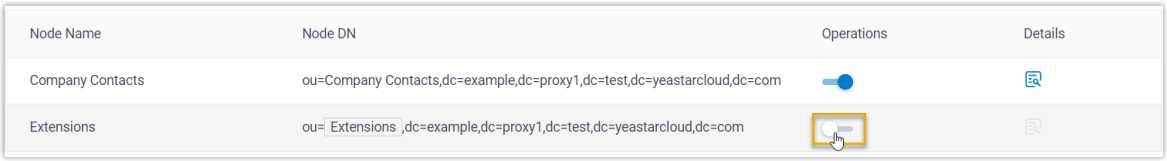


Table 7.

Setting	Description
LDAP Host	The LDAP Server address of Yeastar P-Series Cloud Edition. LDAP Client connects to the LDAP Server via the address.
LDAP Mode	The connection protocol used between the LDAP Server and the LDAP Clients.
LDAP Port	The LDAP port of the LDAP Server.
LDAPs Port	The LDAPs port of the LDAP Server.
Base DN	The Base DN is based on the **LDAP Host** address of Yeastar P-Series Cloud Edition.

4. Click the **LDAP Nodes** tab, enable or disable the nodes according to your needs.



If a node is disabled, you can not query the information under this node.

Result

The Yeastar P-Series Cloud Edition is now working as an LDAP Server. You can store contact information in the PBX directly. Users can connect an IP phone to PBX via LDAP, and query the contact information from IP phone directly.

Set up LDAP Client

Auto Provision LDAP for IP Phones

You can configure the LDAP for IP phone via Auto Provisioning, which is more convenient and easy to operate.

Supported IP phones

This topic can be applied to the IP phones listed in Auto Provisioning - Supported Devices.

Prerequisites

- Make sure the PBX version is 84.6.0.24 or later.
- You have set up the PBX as an LDAP server.
- The phone is connected to Yeastar P-Series Cloud Edition via Auto Provisioning, and has been assigned with an extension.

  For more information, see Auto Provision IP Phones.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**, click to edit the phone.
2. Under **Phone** tab, scroll down to the **LDAP Directory** section, set up the LDAP feature according to your needs.

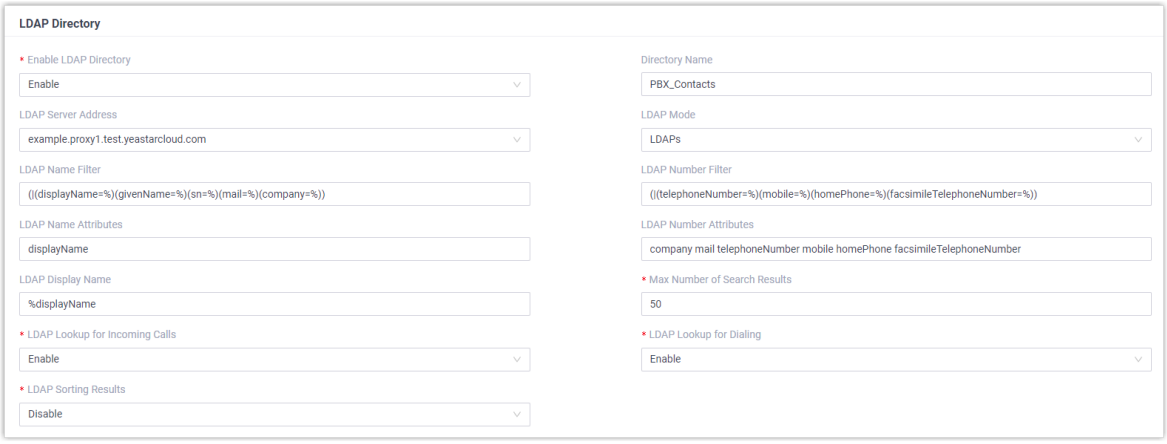


Table 8.

Setting	Description	Example
Enable LDAP Directory	Enable or disable the LDAP directory feature.	Enable
Directory Name	Specify a name for the LDAP directory.	PBX_Contacts
LDAP Server Address	Enter the LDAP Server address of Yeastar P-Series Cloud Edition.	example.proxy1.test.yeastarcloud.com
LDAP Mode	Select the connection mode between the LDAP Server and the IP phone.	LDAPs
LDAP Name Filter	Specify the name attributes for LDAP contact name lookup. **Note:** • The `*` symbol in the filter stands for any character. • The `%` symbol in the filter stands for the entering string used as the prefix of the filter condition.	(|(displayName=%)(givenName=%) (sn=%)(mail=%)(company=%))
LDAP Number Filter	Specify the number attributes for LDAP searching. **Note:** • The `*` symbol in the filter stands for any character. • The `%` symbol in the filter stands for the entering string used as the prefix of the filter condition.	(|(telephoneNumber=%)(mobile=%) (homePhone=%)(facsimileTelephoneNumber=%))
LDAP Name Attributes	Specify the name attributes of each record to be returned by the LDAP Server. The user can configure multiple name attributes separated by space.	displayName
LDAP Number Attributes	Specify the number attributes of each record to be returned by the LDAP Server. The user can configure multiple number attributes.	telephoneNumber mobile homePhone
LDAP Display Name	Specify the display name of the contact record displayed on the LCD screen.	%displayName

Setting	Description	Example
	Note: This parameter must start with % symbol.	
Max Number of Search Results	Specify the maximum number of search results to be returned by the LDAP Server.	50
LDAP Lookup for Incoming Call	Enable or disable IP phone to perform an LDAP search when receiving an incoming call.	Enabled
LDAP Lookup for Callout	Enable or disable IP phone to perform an LDAP search when placing a call.	Enabled
LDAP Sorting Results	Enable or disable IP phone to sort out search results in alphabetical and numerical order.	Enabled

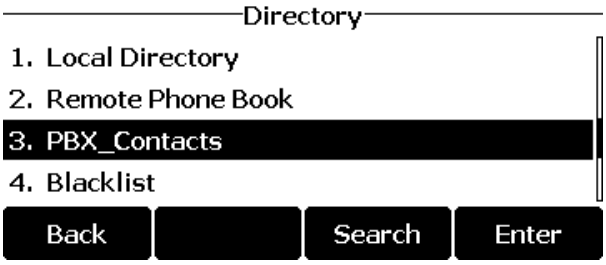
3. Click **Save**.

   The page returns to **Auto Provisioning > Phones**.

4. Click beside the phone to reprovision the settings.
5. In the pop-up dialog box, click **OK**.

Result

You can now query the contact information from IP phone on **Menu > Directory**.



Related information

Auto Provision Function Keys for Phones

Manual Configuration Examples

LDAP Configurations on Yealink Phones

This topic takes the Yealink SIP-T53W IP phone with a firmware version of 93.85.0.5 to describe how to configure LDAP on Yealink IP phones.

Configuration example

The example configurations are set according to default settings of Yeastar P-Series Cloud Edition LDAP Server. You can use the following settings as a starting point and adjust the filter and display attributes according to your needs.

Prerequisites

You have set up the PBX as an LDAP server.

Procedure

1. Log in to the Yealink phone web interface, go to **Directory > LDAP**.
2. Turn on the **LDAP Enable** feature switch, and enter the desired values in the corresponding fields.

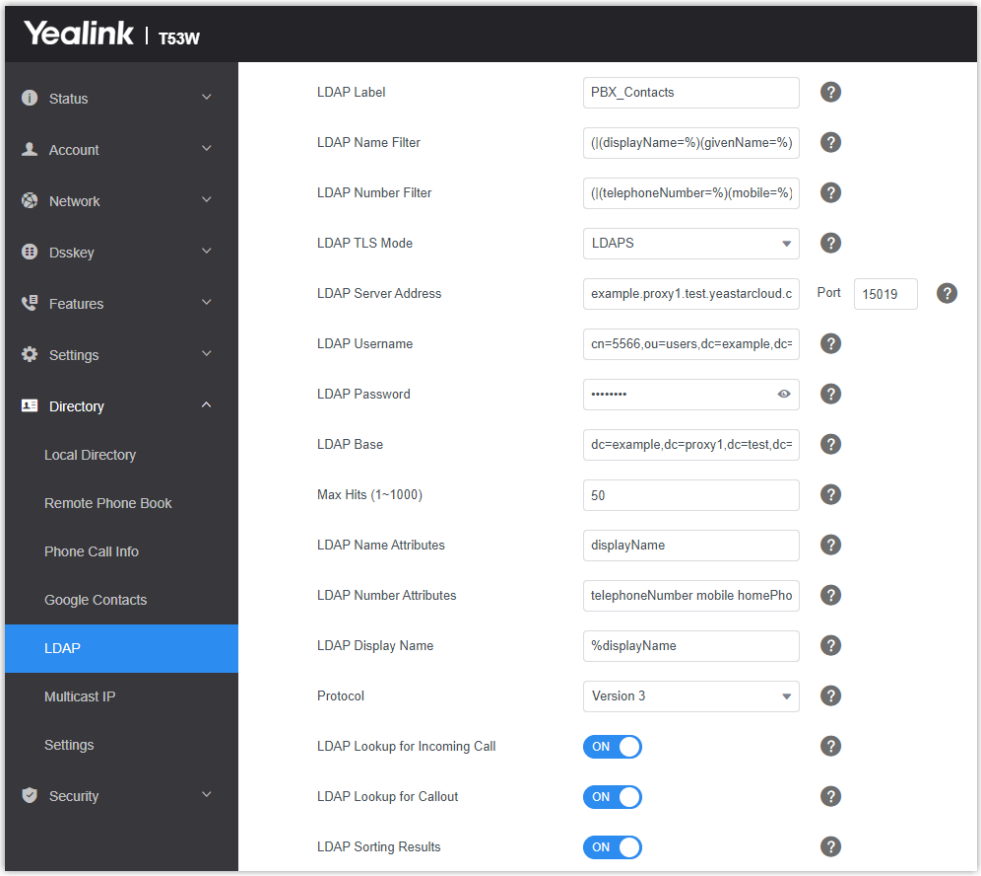


Table 9.

Setting	Description	Example
LDAP Label	Specify the name of LDAP phonebook.	PBX_Contacts
LDAP Name Filter	Specify the name attributes for LDAP contact name lookup. **Note:** • The `*` symbol in the filter stands for any character. • The `%` symbol in the filter stands for the entering string used as the prefix of the filter condition.	(|(displayName=%)(givenName=%) (sn=%)(mail=%)(company=%))

Setting	Description	Example
LDAP Number Filter	Specify the number attributes for LDAP searching. **Note:** • The `*` symbol in the filter stands for any character. • The `%` symbol in the filter stands for the entering string used as the prefix of the filter condition.	(|(telephoneNumber=%)(mobile=%)(homePhone=%)(facsimileTelephoneNumber=%)))
LDAP TLS Mode	Specify the connection mode between the LDAP Server and the IP Phone.	LDAPS
LDAP Server Address	Enter the LDAP Server address of Yeastar P-Series Cloud Edition.	example.proxy1.test.yeastarcloud.com
Port	Enter the LDAP Server port.	15019
LDAP Username	Enter the username to log in to the LDAP Server. **Note:** Obtain the username from PBX on **Contacts > LDAP Server > LDAP Credentials > LDAP Account Username**.	cn=5566,ou=users,dc=example,dc=proxy1,dc=test,dc=yeastarcloud,dc=com
LDAP Password	Enter the password to log in to the LDAP Server. **Note:** The password is the registration password of the user extension.	Regpwd123
LDAP Base	Enter the **Base DN** obtained from PBX, which is used as the LDAP search base.	dc=example,dc=proxy1,dc=test,dc=yeastarcloud,dc=com

Setting	Description	Example
Max Hits (1~1000)	Specify the maximum number of search results to be returned by the LDAP Server.	50
LDAP Name Attributes	Specify the name attributes of each record to be returned by the LDAP Server. The user can configure multiple name attributes separated by space.	displayName
LDAP Number Attributes	Specify the number attributes of each record to be returned by the LDAP Server. The user can configure multiple number attributes.	telephoneNumber mobile homePhone
LDAP Display Name	Specify the display name of the contact record displayed on the LCD screen. **Note:** This parameter must start with % symbol.	%displayName
Protocol	The LDAP protocol version. Yeastar P-Series Cloud Edition uses Version 3.	Version 3
LDAP Lookup for Incoming Call	Enable or disable IP phone to perform an LDAP search when receiving an incoming call.	Enabled
LDAP Lookup for Callout	Enable or disable IP phone to perform an LDAP search when placing a call.	Enabled
LDAP Sorting Results	Enable or disable IP phone to sort out search results in alphabetical and numerical order.	Enabled

3. Click **Confirm** to apply the changes.

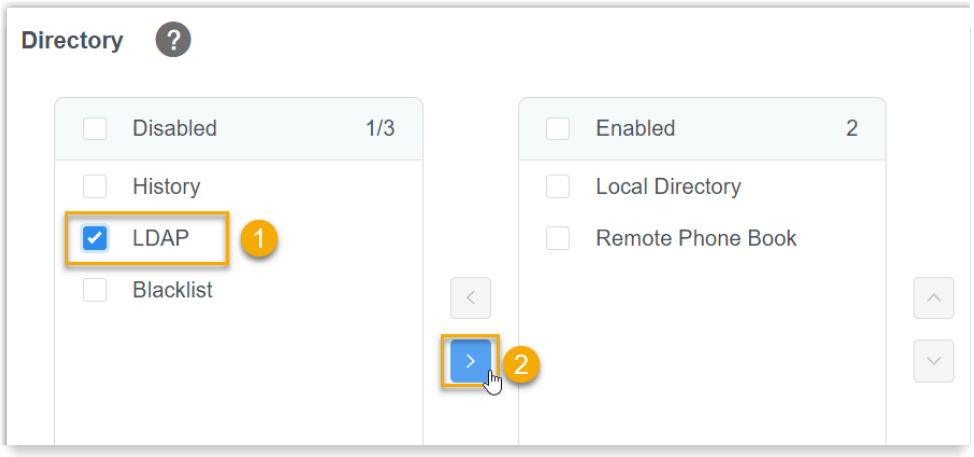
Result

Now you can directly check the contact information stored in the PBX from the IP phone.

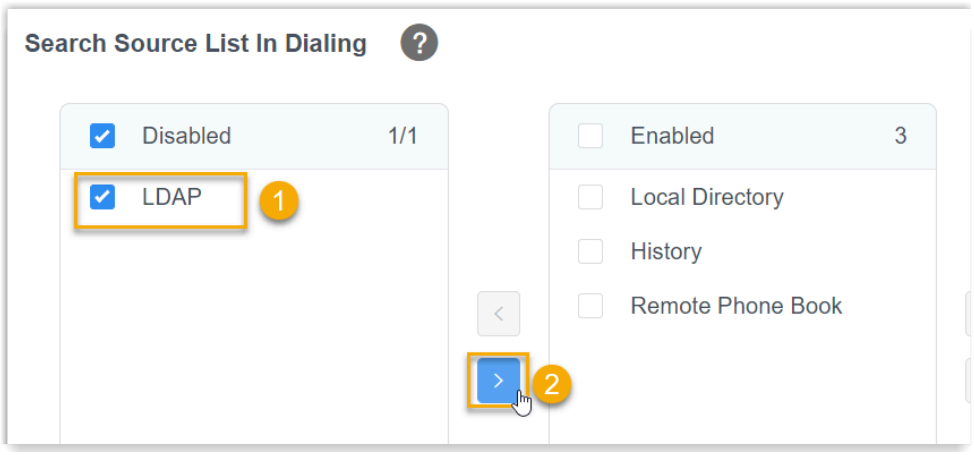
Search contacts via Directory

Enable LDAP directory on Yealink phone

1. Log in to the Yealink phone web interface, go to **Directory > Settings**.
2. In the **Directory** section, add **LDAP** from the **Disabled** box to the **Enabled** box.



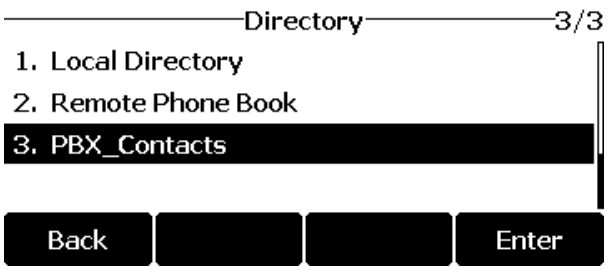
3. **Optional:** In the **Search Source List In Dialing** section, add **LDAP** from the **Disabled** box to the **Enabled** box.



4. Click **Confirm**.

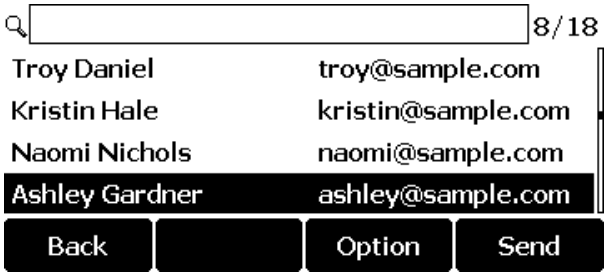
Search LDAP Contacts

1. On the IP phone, press **Directory** and enter the LDAP phonebook.



2. Search the contact name or number using the keypad.

   The contacts whose name or phone number matching the characters entered will appear on the phone screen.

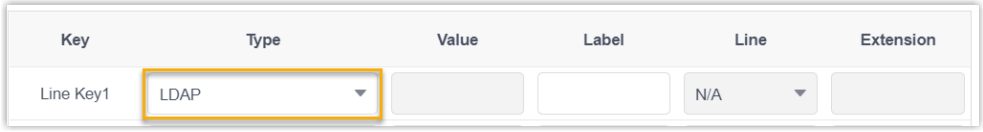


3. Press the navigation key to select the desired contact.
4. Press **Send** to call the contact.

Search contacts via LDAP key

Set an LDAP Key on Yealink Phone

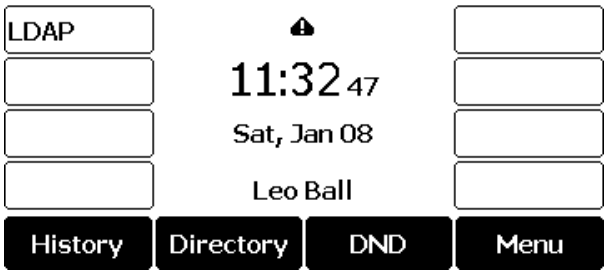
1. Log in to the Yealink phone web interface, go to **DssKey > Line Key** to configure a line key.
2. In the drop-down list of **Type**, select **LDAP**.



3. Click **Confirm**.

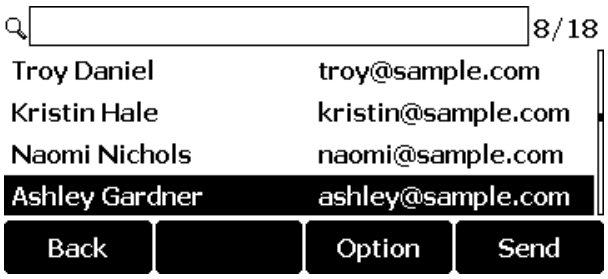
Search LDAP Contacts

1. Press the LDAP key to access the LDAP phonebook.



2. Search the contact name or number using the keypad.

   The contacts whose name or phone number matching the characters entered will appear on the phone screen.



3. Press the navigation key to select the desired contact.
4. Press **Send** to call the contact.

Organization

Organization Overview

Organizational structure is the group of rules, roles, relationships, and responsibilities that outline how your company's activities are directed to meet its goals. Yeastar P-Series Cloud Edition provides Organization feature to help you organize employees by department based on their specific skills and corresponding function in the company, and enjoy easier administration with department-level control.

Organization vs Extension Group

Yeastar provides **Organization** feature and **Extension Group** feature to help you categorize and manage extensions.

The following contents compare the advantages and differences between the two features.

Organization

Organization feature is used to define a hierarchy within a company, ideal for large companies with many departments and for those companies that attach more importance to separation of duties.

Organization feature helps you achieve the followings:

- Multi-layer departments, displayed in hierarchical tree structure.
- Flexible adjustments for departments, adapting to a changing business environment.
- Clear-cut reporting structure, clarifying the reporting relationships across the company and every individual's role and responsibilities.

Extension Group

Extension Group feature is used to categorize extensions with common function or purpose into the same group, ideal for companies that attach more importance to call management.

Extension Group feature helps you achieve the followings:

- Same-layer groups, displayed in strict alphabetical order.

- Group permission presets, realizing granular control over users' call permission.
  - Preset permission on a per group basis: Preset different permissions for groups with different calling needs.
  - Preset permission on a per user role basis: Split users into Manager/User/Custom roles and implement role-based permission assignment.

For more information about extension group, see Extension Group Overview.

We provide the following figures to visualize the difference between Organization and Extension Group in display:



Organization application

After you enable Organization feature and set up departments, the followings can be achieved:

- On Linkus clients, extension users can search for and find colleagues by departments.

> **Note:**
> Make sure Linkus clients meet the following version requirements:
> - Linkus iOS version: 4.8.5 or later.
> - Linkus Android version: 4.8.6 or later.

- Linkus Web Client: 84.7.0.17 or later.

- On PBX web portal, you can implement department-based control over users' permission:
  - Control the visibility to specific extensions or company contacts.
  - Control the access to all the call features and Call Center Console.

> **Note:**
> The access to Operator Panel is under the control of Extension Group, be the Organization feature enabled or not.

Related information

Enable or Disable Organization Management

Enable or Disable Organization Management

You can enable or disable organization management feature based on your plan for company structure.

Enable organization management

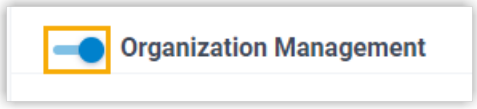
To group extension users into departments, you need to enable the **Organization Management** feature.

Prerequisites

The version of Yeastar P-Series Cloud Edition is 84.7.0.17 or later.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Preferences**.
2. Turn on the option **Organization Management**.



3. In the **Company Name** field, enter your company name. The name will be used as the root organization.

> **Note:**

> If you have set up company information, the pre-defined company name is automatically synchronized here.

4. Click **Save** and **Apply**.

Result

The **Organization Management** feature is enabled.

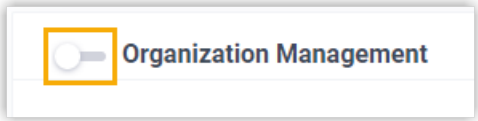
What to do next

Create departments.

Disable organization management

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Preferences**.
2. Turn off the option **Organization Management**.



3. Click **Save** and **Apply**.

Result

The **Organization Management** feature is disabled, which bring changes to the way that extensions are displayed and permissions that extensions have.

- On PBX web portal, the organizational tree and organizational configuration page are hidden; On Linkus clients, extension users are arranged in extension groups.
- Extension users have no access to the features that are granted to organizations.

Set up Organizations

Organizational structure helps your company stay organized, improve communication and collaboration productively. This topic describes how to set up organizations.

Limitation

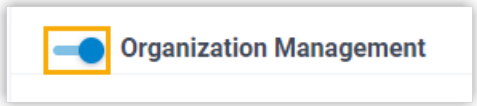
Maximum Number of Extensions (N)	N ≤50	50 < N ≤ 200	N > 200
Layers of Departments	15	15	15
Number of Departments	50	200	500

Prerequisites

The version of Yeastar P-Series Cloud Edition is 84.7.0.17 or later.

Step 1. Enable Organization Management

1. Log in to PBX web portal, go to **PBX Settings > Preferences**.
2. Turn on the option **Organization Management**.



3. In the **Company Name** field, enter your company name. The name will be used as the root organization.

> **Note:**
> If you have set up company information, the pre-defined company name is automatically synchronized here.

4. Click **Save** and **Apply**.

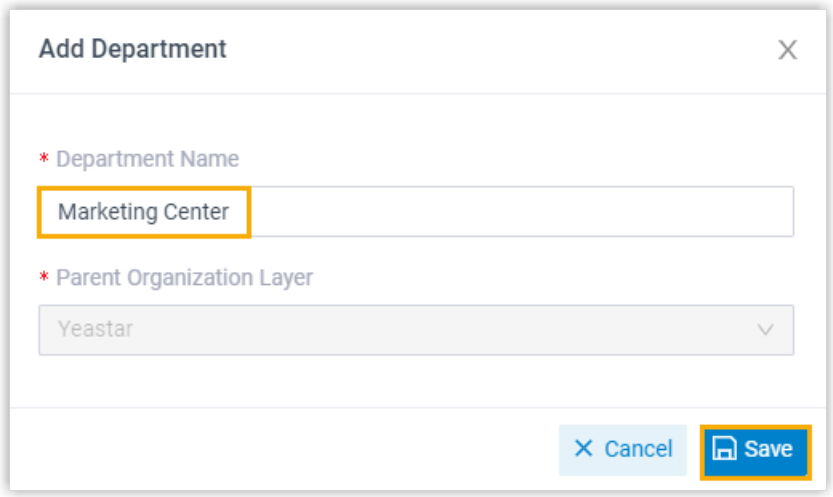
Step 2. Create departments

1. Go to **Extension and Trunk > Extension > Organization**.

   The root organization (namely the company name) is displayed.



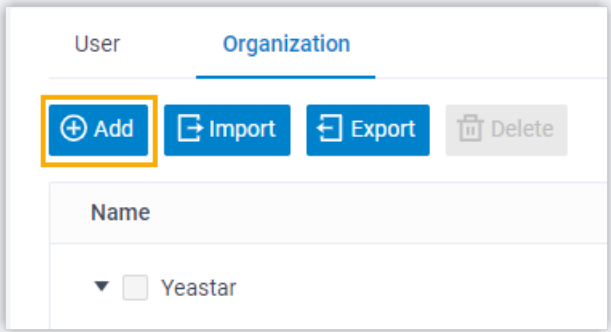
2. Click + beside the root organization.
3. In the pop-up window, configure the following information, then click **Save**.



- **Department Name**: Enter a department name.
- **Parent Organization Layer**: The root organization is automatically filled in.

Result

- The department is created. You can create more departments as the instructions provided above. In this way, the parent organization is auto filled instead of manually selected.

> **Tip:**
> To select parent organization at you will when creating departments, you can click **Add** to create departments.
>
> 
>

- On Linkus clients, users can see all the departments. To restrict users from viewing specific departments, see Set up Extension Visibility.

What to do next

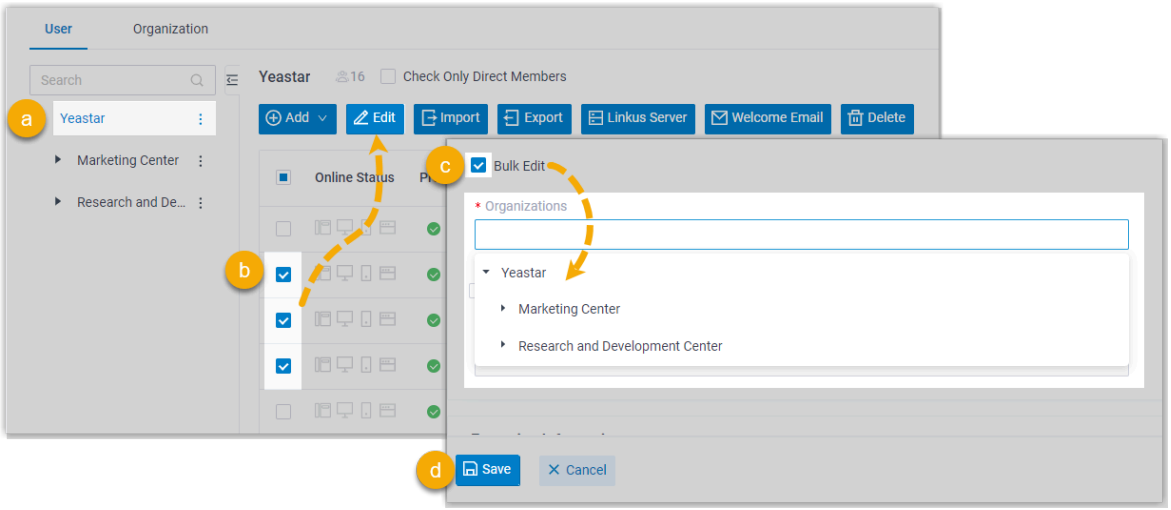
Add Users to Organizations.

Add Users to Organizations

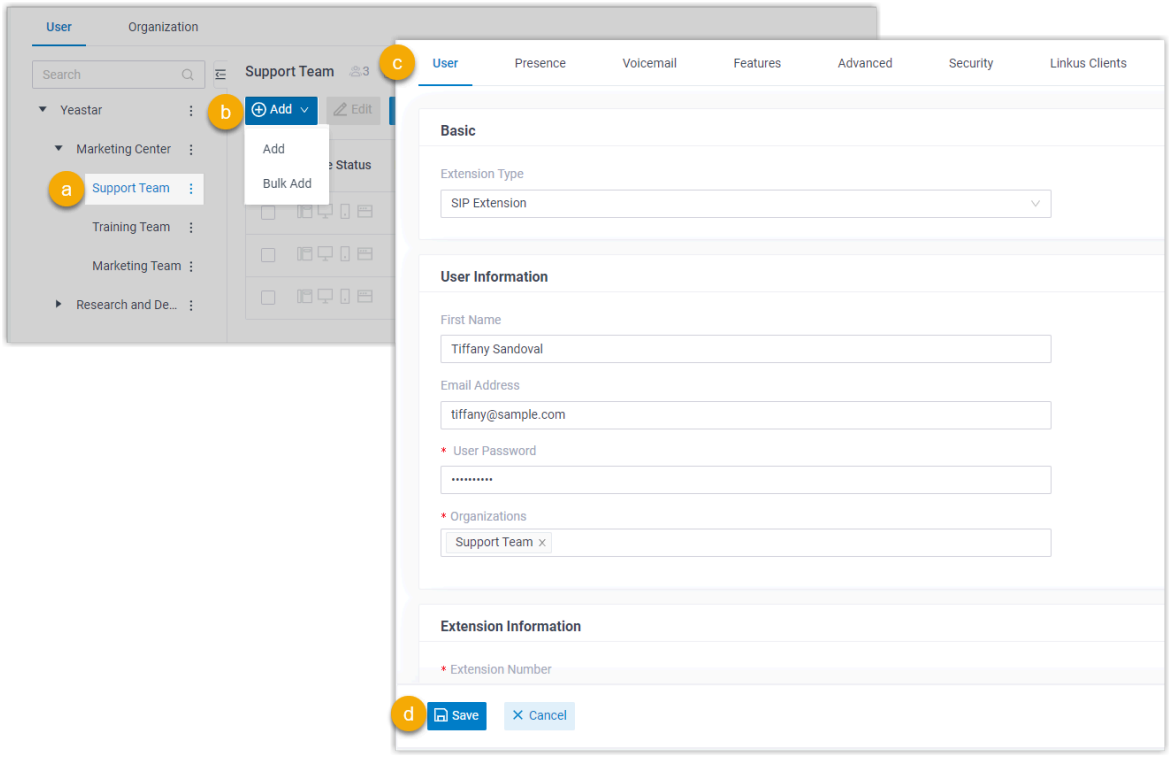
After setting up organizations, you need to group users into departments. This topic describes how to add users to departments.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > User**.
2. To add existing users to an organization, do as follows:



   a. On the left organizational tree, click root organization.
   b. On user list, select the checkboxes of desired extensions, then click **Edit**.
   c. In the **User Information** section, select the checkbox of **Bulk Edit** for organizations, then select desired departments.
   d. Click **Save** and **Apply**.
3. To add new users to an organization, do as follows:



a. On the left organizational tree, click a department.
b. Click **Add**, then select **Add** to add an extension.
c. Configure the extension as needed.

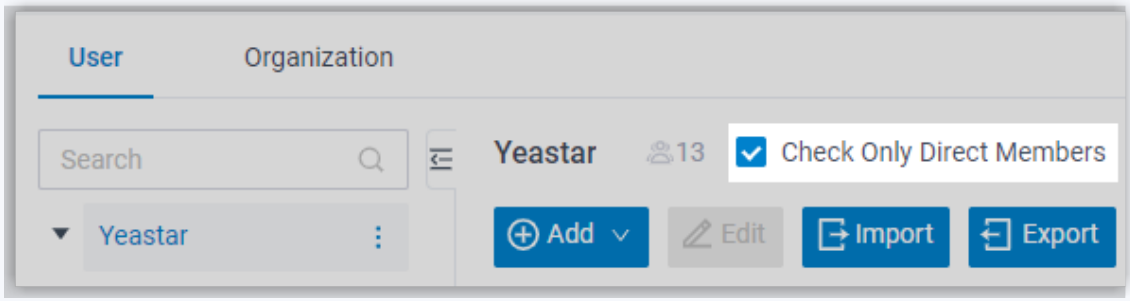
> **Note:**
> **Organization** is auto filled with the one that you have selected.

d. Click **Save** and **Apply**.

Result

Users are added to the specified departments. You can click a department to check all the associated members.

> **Note:**
> By default, when you click on a department, all the users within the department are displayed, be they belong to the parent department or the sub-departments. To hide the users of sub-departments, select the checkbox of **Check Only Direct Members**.



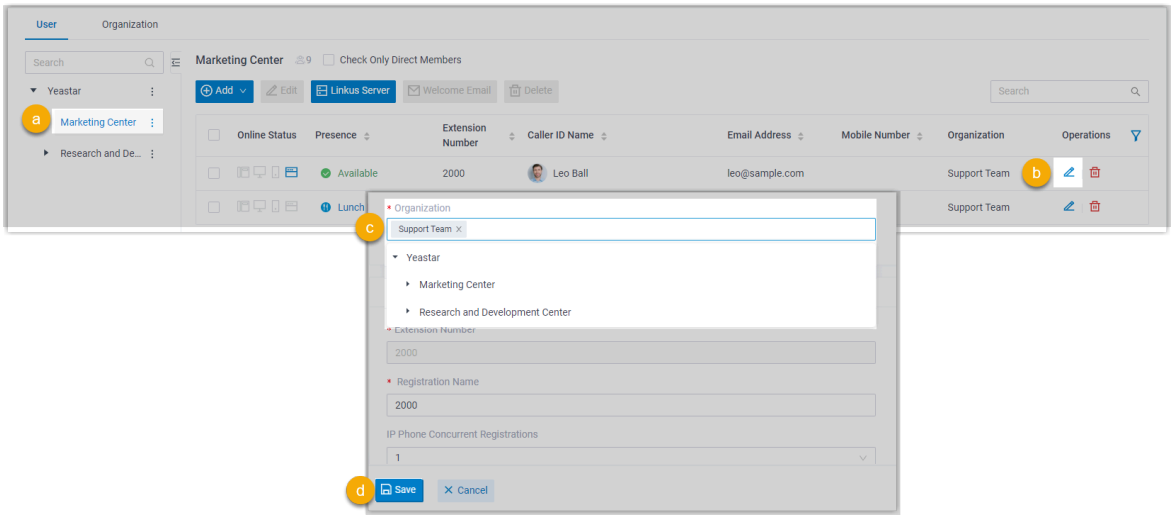
Manage Users within Organizations

This topic describes how to change users' departments or remove users from departments when there are job changes of employees.

Change users' departments

You can change users' departments when some of them transfer jobs at your company.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > User**.
2. To change a user's department, do as follows:



   a. On the left organizational tree, click a desired department.

      All the extensions within the department are displayed.

   b. Click beside a desired extension.

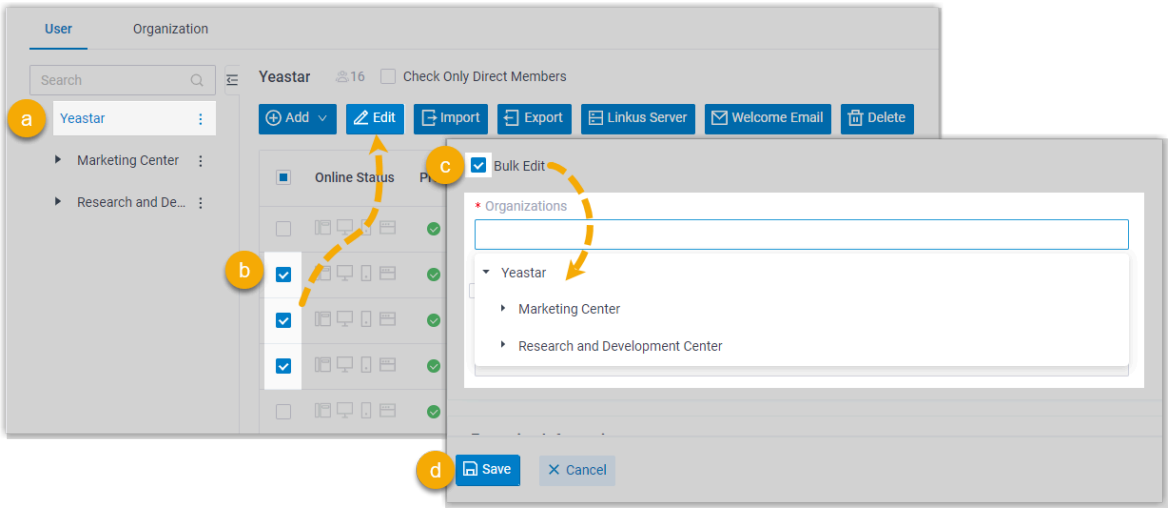
   c. In the **Organization** field , change department as needed.

> **Note:**
> An extension must be associated with at least one department.

d. Click **Save** and **Apply**.

3. To change multiple users' departments, do as follows:

> **Note:**
> This is suitable for changing multiple users to the same department. To change multiple users to different departments, you need to proceed one by one as step2 instructs.



a. On the left organizational tree, click the root organization.

   All the extensions within the organization are displayed.

b. Select the checkboxes of desired extensions, then click **Edit**.

   The departments to which the extensions belong are cleared.

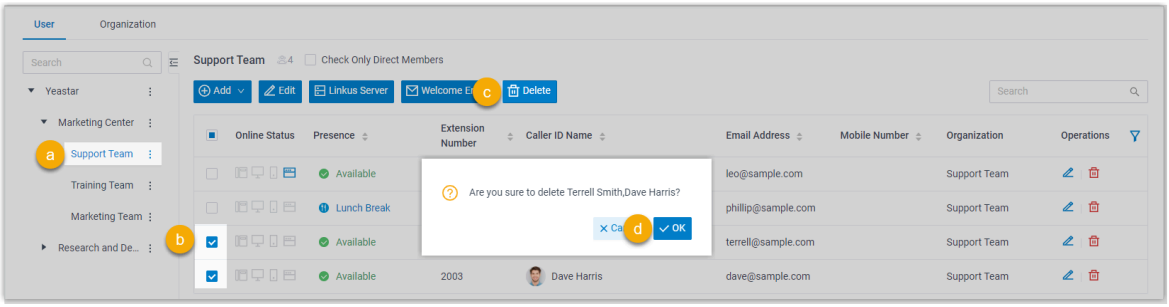
c. Select the checkbox of **Bulk Edit** for organization, then reselect departments.

d. Click **Save** and **Apply**.

Remove users from departments

You can remove users from departments when some of them leave their jobs.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > User**.
2. To remove users from the same department, do as follows:



a. On the left organizational tree, click a desired department.

   All the extensions within the department are displayed.

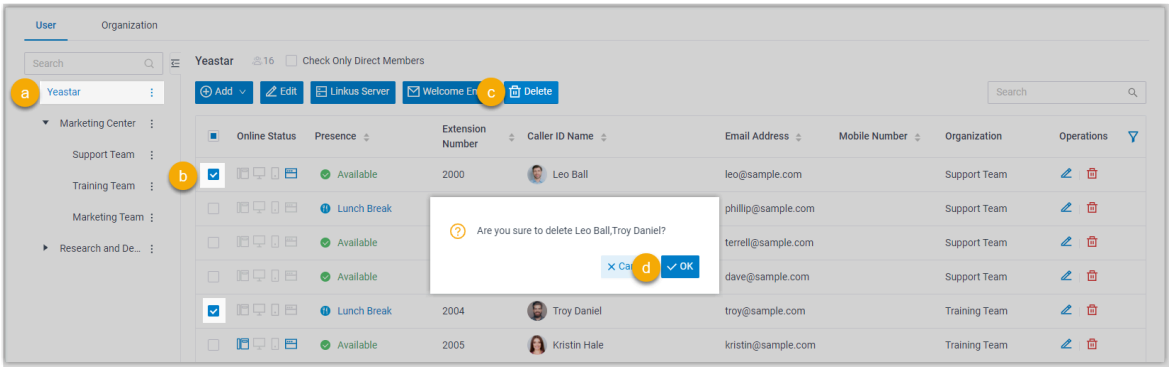
b. Select the checkboxes of desired extensions, then click **Delete**.

c. In the pop-up window, click **OK**.

d. Click **Apply**.

   The selected extensions are deleted from the system.

3. To remove users from different departments, do as follows:



a. On the left organizational tree, click the root organization.

   All the extensions within the organization are displayed.

b. Select the checkboxes of desired extensions, then click **Delete**.

c. In the pop-up window, click **OK**.

d. Click **Apply**.

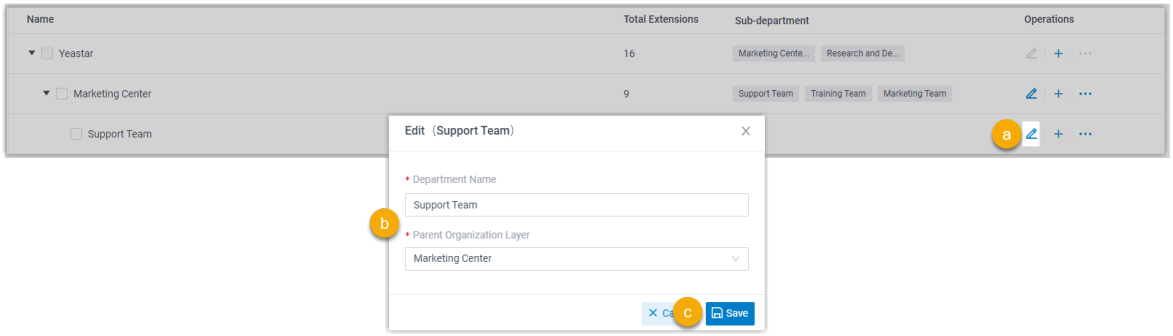
   The selected extensions are deleted from the system.

Manage Organizations

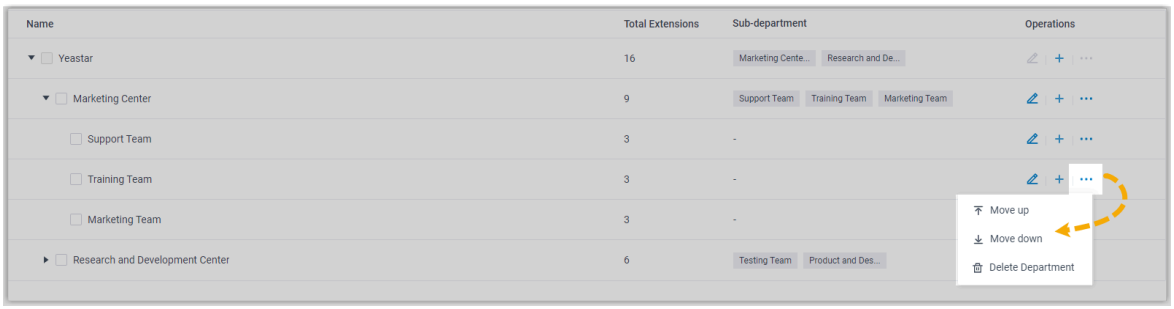
To remain competitive or adapt to changes in the company, you may change organizational structure. This topic describes how to manage the organizations on PBX web portal.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > Organization**.
2. To rename department or change parent organization, do as follows:



   a. Click ✎ beside the desired department.
   b. In the pop-up window, rename department or change parent organization layer.
   c. Click **Save**.
3. To adjust the order of departments, click ..., then select **Move Up** or **Move Down** to adjust the order.

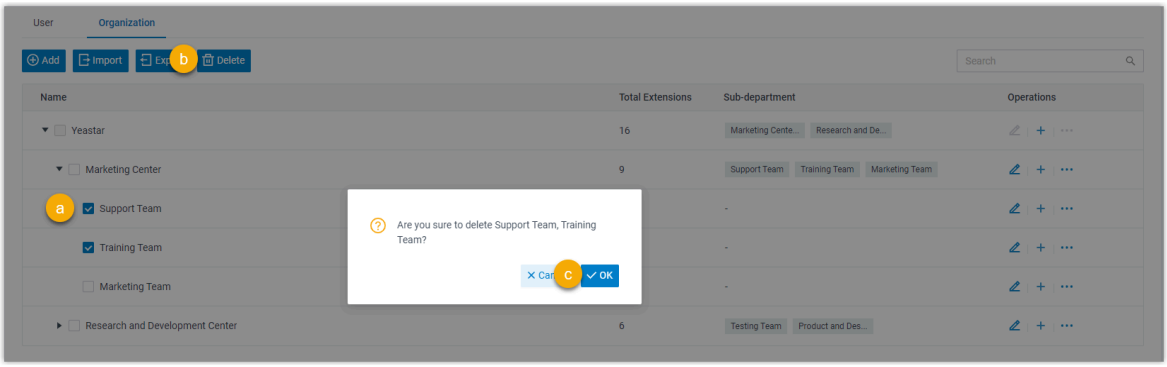


   On PBX web portal and Linkus clients, departments are displayed in the new order.
4. To delete departments, do as follows:

> **Note:**
> - If there are sub-departments under the departments that you want to delete, you need to delete the sub-departments first.
> - After you delete departments, the extensions within the departments will not be deleted, but they have no access to the features that are granted to the departments, and they will be grouped into root organization if they only belong to the departments deleted.

   a. Select the checkboxes of desired departments, then click **Delete**.

b. In the pop-up window, click **OK**.



Export and Import Organizations

The organizations configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired organization information in the exported file, and import the file to PBX again. This topic describes how to export and import organizations.

Export all organizations

You can export all the organizations to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > Organization**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Organization Parameters.

Import organizations

We recommend that you export organization data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8 .CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters

- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Organization Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension > Organization**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, select your CSV file.
4. Click **Import**.

   The organization data in the CSV file will be displayed in the **Organization** list.

Related information

Import and Export -FAQ

Extension Group

Extension Group Overview

Yeastar P-Series Cloud Edition supports to add specific extensions to a group, assign user types to these extensions, and grant permissions to extension users with different user types.

What is Extension Group

Extension group is a group that contains a number of extensions with a common function or purpose. Extension group is displayed on Linkus clients, which allows users to easily find a colleague within a group, and makes it possible for authorized users to control calls of members within a specific group on Linkus Web Client.

User types in an extension group

A user type is a permission set, which allows you to control users' access to and usage of Operator Panel. Yeastar P-Series Cloud Edition provides 3 user types. You can grant permissions to each user type and assign user types to group members.

Default user types

- **Manager**: Assign the user type to a leader, so that he or she can manage members' calls.
- **User**: Assign the user type to ordinary members. Any time you add members to a group, they are assigned with the user type by default.

The following table displays default permissions for **Manager** and **User**, you can change the permissions according to your needs. For more information, see View or change permissions for managers and users.

Module	Permission	Manager	User
Operator Panel	Switch group members' presence	√	×
	Call distribution management (Redirect, Transfer, Drag and Drop operation)	√	×
	Pick up or hang up other extensions' calls	√	×

Module	Permission	Manager	User
	Call monitoring operations (Listen, Whisper, Barge-in)	√	×
	Call parking operations (Park, Retrieve)	√	×
	Route calls directly from IVR regardless of the IVR menu	√	×
	Switch Business Hours and Holidays status	×	×
	Switch extension's recording status	×	×
	Show Company Contacts Matching Results **Note:** Once enabled, the contacts matching results will be displayed on Operator Panel call panel, regardless of users' visibility permission for company contacts.	×	×

Custom user type

Custom: If you want to grant permissions to a specific member, you can assign the user type to a desired member, and customize permissions.

For more information, see View or change permissions for a member with custom user type.

Default extension group

Yeastar P-Series Cloud Edition has a built-in group **Default_All_Extensions** that contains all the extensions on the PBX. Any time you create an extension, the extension will be automatically added to the extension group. You can delete the group, or create one or more groups according to your needs.

For more information, see Create an Extension Group.

Create an Extension Group

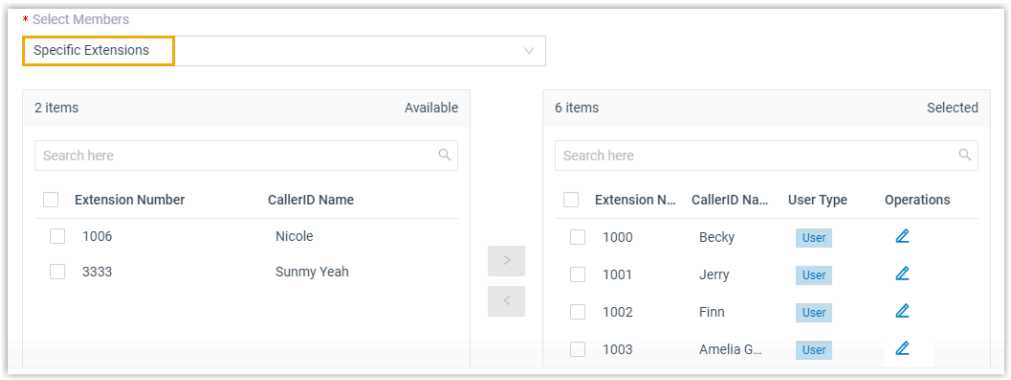
This topic describes how to create an extension group.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, click **Add**.
2. Configure basic settings for the extension group.
   a. In the **Name** field, enter a group name to help you identify it.
   b. In the **Select Members** drop-down list, set which extensions will be added to the group.
      - **All Extensions**: If you choose the option, all the extensions will be moved to the **Selected** box.

        > **Note:**
        > ONLY one group that contains all the extensions is allowed.

      - **Specific extensions**: If you choose the option, select the desired extensions from **Available** box to **Selected** box.



   c. Assign user types for group members.

      > **Note:**
      > Users of different user types have different permissions. For more information, see User types in an extension group.

      i. In the **Selected** box, click ✎ beside the desired member.
      ii. In the pop-up window, configure the **User Type** and permissions.

- If you select **Manager** or **User**, the member has all the permissions that are granted to the user type.

  > **Note:**
  > The permissions of **Manager** and **User** are pre-defined. To change the permissions, see View or change permissions for managers and users.

- If you select **Custom**, select the checkboxes of the desired permissions.

iii. Click **Save**.

3. Click **Save**.

Result

- The extension group is displayed on **Extension Group** list.
- No one can view the group on Linkus clients. To allow specific users to view the group, see Set up Extension Visibility.

Manage Extension Groups

This topic describes how to edit or delete extension groups.

Edit an extension group

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, click beside the desired group.
2. Change group settings according to your needs.
3. Click **Save** and **Apply**.

Delete extension groups

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**.
2. To delete an extension group, do as follows:
   a. Click beside the desired group.
   b. In the pop-up dialog box, click **OK**.
   c. Click **Apply**.

3. To delete extension groups in bulk, do as follows:
    a. Select the checkboxes of the desired groups, click **Delete**.
    b. In the pop-up dialog box, click **OK**.
    c. Click **Apply**.

    The groups are removed from **Extension Group** list and are not displayed on Linkus clients.

Assign a User Type to a Group Member

Members of different user types have different permissions. You can control members' access to specific features by assigning different user types in an extension group. This topic describes how to assign a user type to a group member.

Assign a default user type to a group member

Yeastar P-Series Cloud Edition provides two default user types: **Manager** and **User**, each of them has preset permissions. By assigning the two user types to members, you can bulk grant permissions to multiple members who share common responsibilities.

Prerequisites

Familiarize yourself with permissions of **Manager** and **User** in the desired group and change permissions according to your needs.

For more information, see View or change permissions for managers and users.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, edit the desired extension group.
2. Assign **Manager** or **User** to a group member.
    a. In the **Members** section, click  beside the desired member.
    b. In the **User Type** drop-down list, select **Manager** or **User** according to your needs.
    c. Click **Confirm**.

Result

The member's user type and permissions in the group are updated.

Assign a custom user type to a group member

If you want a member to have different permissions from members with default user types, you can assign a custom user type to a desired member, and customize permissions.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, edit the desired extension group.
2. Assign **Custom** to a group member, and grant permissions to the member according to your needs.
    a. In the **Members** section, click  beside the desired member.
    b. In the **User Type** drop-down list, select **Custom**.
    c. Select the checkboxes of the desired permissions.
    d. Click **Confirm**.
3. Click **Save** and **Apply**.

Result

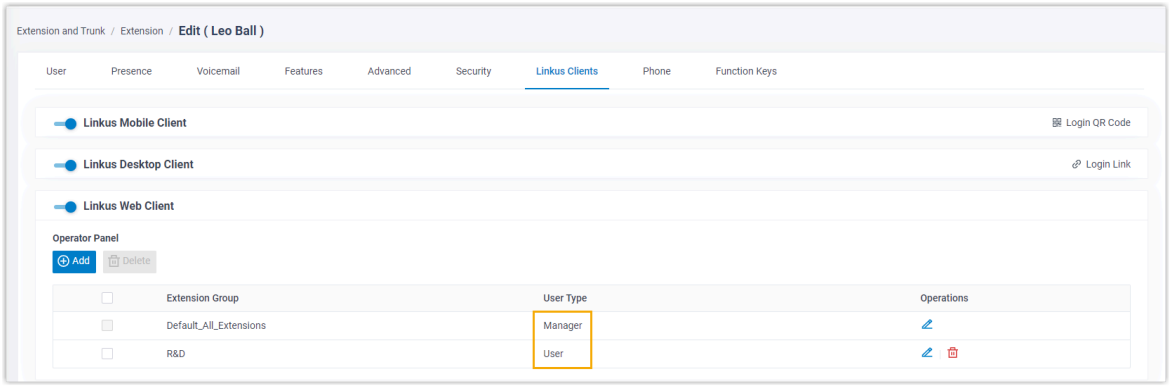
The member's user type and permissions in the group are updated.

View or Change a Member's User Type in Multiple Groups

If an extension user plays different roles in different extension groups, you can quickly view or change multiple user types of the extension user without having to go to each group to view or assign the user types. This topic describes how to view or change a member's user type in multiple groups.

View a member's user type in multiple groups

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click  beside desired extension.
2. Click **Linkus Clients** tab.
3. In the **Operator Panel**section, you can see all the groups to which the extension user belongs. Check the user's user type in each group in **User Type** column.

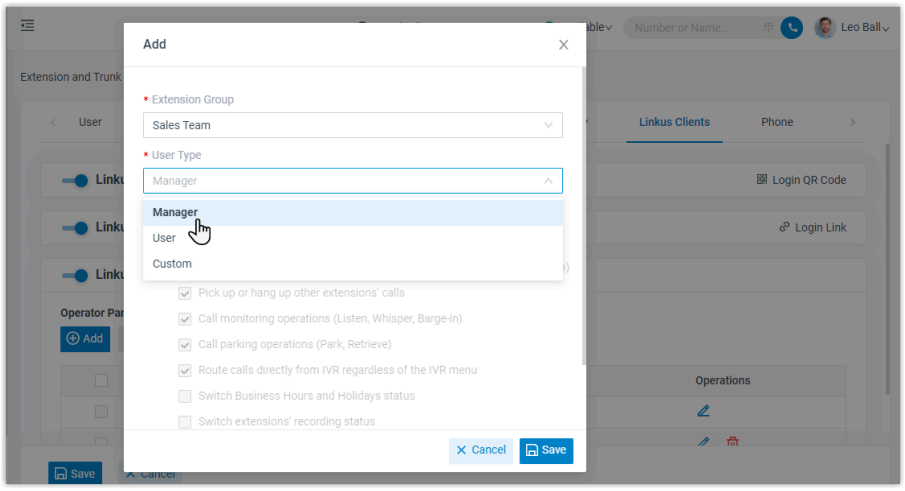


Change a member's user type in multiple groups

The permissions of **Manager** and **User** vary from one group to another. Make sure you change permissions for the right group.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click beside desired extension.
2. Click **Linkus Clients** tab.
3. In the **Operator Panel** section, change the extension user's user type in a group.
    a. Click beside the desired extension group.
    b. In the **User Type** drop-down list, select a user type.
        - If you select **Manager** or **User**, the user has all the permissions that are granted to the user type.
        - If you select **Custom**, select the checkboxes of the desired permissions.



c. Click **Confirm**.

4. Repeat **Step4** to assign user types for the extension in more groups.
5. Click **Save**.

Result

The user's user types and permissions in different groups are updated accordingly.

Related information

Assign a User Type to a Group Member

View or Change Permissions for Group Members

This topic describes how to view or change permissions for group members.

View or change permissions for managers and users

If members are assigned **Manager** or **User** in a group, all the members with the same user type have the same permissions. You can view the permissions of managers and users within a specific group, and change permissions according to your needs.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, edit the desired extension group.
2. Click **Group Permissions** tab.

You can view the permissions that are granted to **Manager** and **User** in the group.

3. To change permissions, do as follows:
    a. Select or unselect the checkboxes of corresponding permissions for **Manager** and **User**.
    b. Click **Save** and **Apply**.

Result

The permissions of all the managers and users in the group are updated in a batch.

What to do next

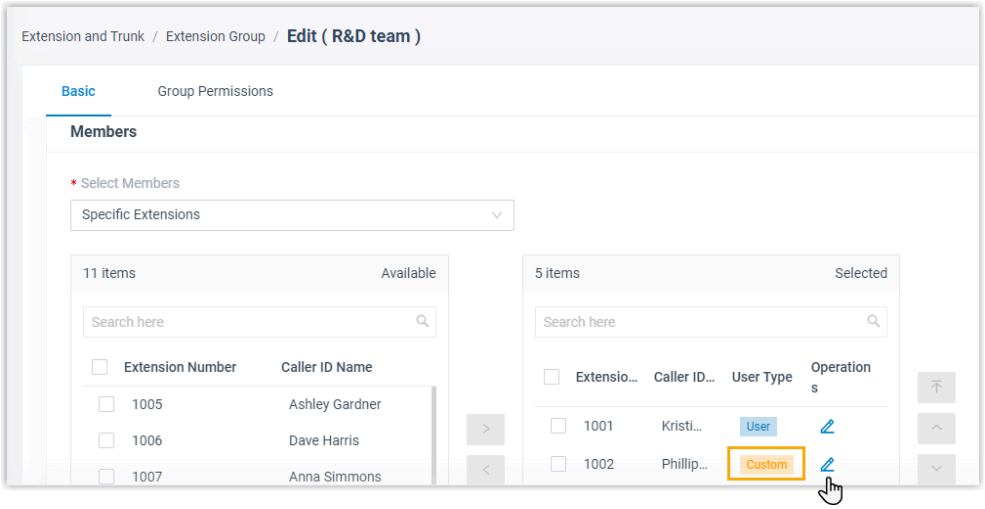
If you want to change members' user types to **Manager** or **User** in the group, see Assign a default user type to a group member.

View or change permissions for a member with custom user type

For members with **Custom** user type assigned, permissions may vary from one member to another. You can view or change permissions for a specific member according to your needs.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**, edit the desired extension group.
2. In the **Members** section, click  beside the desired member whose user type is **Custom**.



a. In the pop-up window, select or unselect the checkboxes of the desired permissions for the user.
b. Click **Confirm**.

3. Click **Save** and **Apply**.

Result

The member's permissions are updated.

Auto Provisioning

Auto Provisioning Overview

Auto Provisioning is a time-saving feature that helps you to manage and deploy IP phones centrally on Yeastar P-Series Cloud Edition. The process of configuring and managing IP phones is simplified, which makes deployment and management of devices fast and convenient.

Auto Provisioning supported devices

Yeastar P-Series Cloud Edition supports various models for Auto Provisioning.

Find the Auto Provisioning - Supported Devices before you start deploying devices.

Auto Provisioning methods

Yeastar P-Series Cloud Edition supports an RPS method to help you provision your IP phones.

RPS

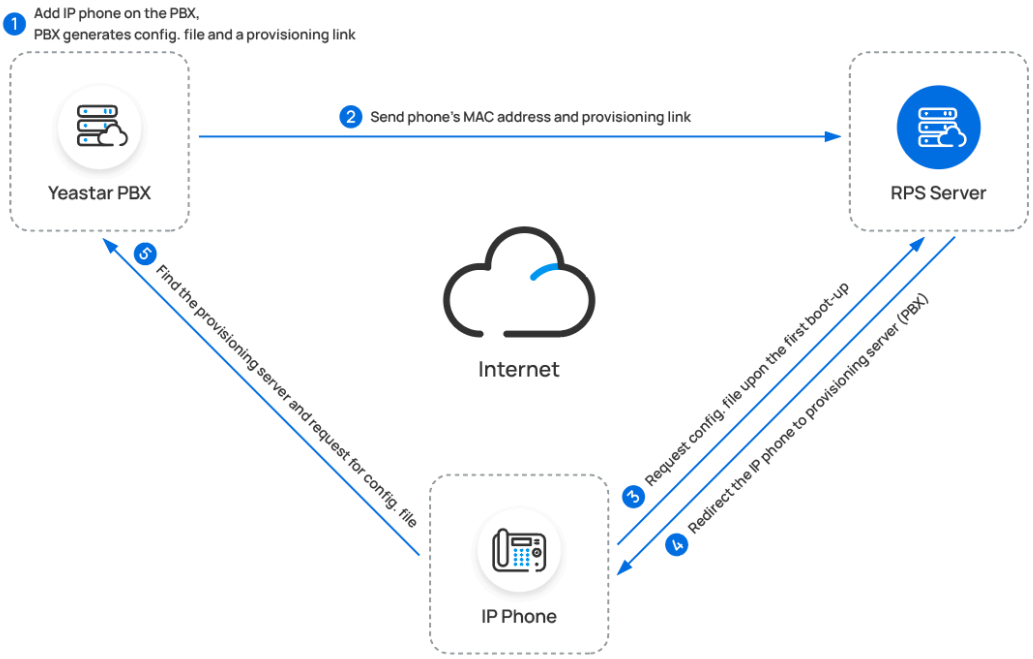
RPS method supports auto provisioning remote IP phones

For more information, see Auto Provision IP Phones.

How Auto Provisioning works

This section introduces how the RPS Auto Provisioning method works with IP phones and the PBX, which can help you understand the operating principle and locate the Auto Provisioning problem rapidly.

RPS Provisioning



Related information

IP Phone Configuration Guides

Provision IP Phones

Auto Provision IP Phones

This topic introduces how to deploy and update IP phones via auto provisioning, which can greatly save time and cost in mass deployment.

> **Note:**
> This topic describes how to provision an IP phone and assign a user's extension to the phone. If you want to set up a hot desking phone via auto provisioning, see Set up a Hot Desking Phone.

Supported IP phones

This topic can be applied to all the IP phones listed in Auto Provisioning - Supported Devices.

Prerequisites

- Make sure that you have downloaded the template for the desired phone model (Path: **Auto Provisioning > Resource Repository > Default Templates**).
- Gather information of IP phone, including Vendor, Model, and MAC address.

Procedure

Step 1. Generate configuration file for an IP phone on the PBX

Step 2. Reboot the IP phone to obtain the configuration file

Step 1. Generate configuration file for an IP phone on the PBX

1. RESET the phone if it is previously used.
2. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
3. Click **Add** to add an IP phone.
4. In the **IP Phone** section, configure phone information as follows:
   - **Vendor**: Select a phone vendor.
   - **Model**: Select a phone model.
   - **MAC Address**: Enter MAC address of the phone.
5. In the **Options** section, configure the following settings.
   - **Template**: Select a desired template from the drop-down list.

     Note:
     The template provides configurations except extension assignment. You can select the default template corresponding to the phone model, or customize your own template. For more information, see Create a Custom Auto Provisioning Template.

   -
   - **Provisioning Link**: A provisioning server URL is generated automatically and displayed on the web page.
   - **Authentication for the First-time Auto Provisioning**: If enabled, users are requested to fill in authentication information on the IP phones before triggering the first-time provisioning.

     Note:
     We recommend that you keep this option selected for security purpose.
6. In the **Assign Extension** section, assign an extension to the phone.

> **Tip:**
> If your desired extension is not listed in the drop-down list, you can check if the extension has been associated with other IP phone.
> - To release the previous phone, see Release an Extension from a Provisioned IP Phone.
> - To associate an extension with multiple IP phones, see Allow Multiple Registrations for One Extension Number.

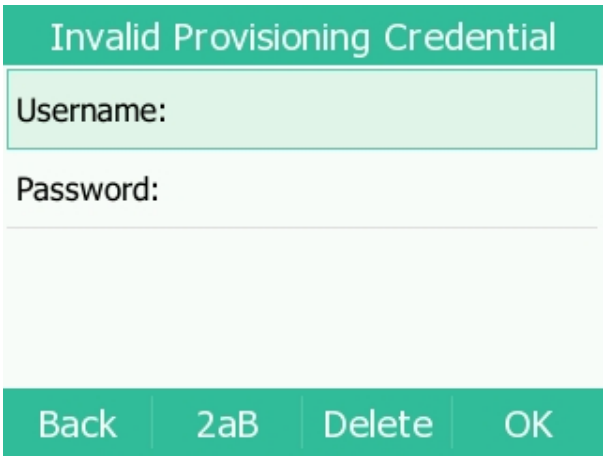
7. Click **Save**.

   A configuration file for the phone is generated in the PBX.

   The PBX will send an event notification of **RPS Request Success**, which means that the phone MAC is added automatically to the RPS server.

Step 2. Reboot the IP phone to obtain the configuration file

1. Reboot the IP phone.
2. If you have enabled **Authentication for the First-time Auto Provisioning** on the PBX, enter the authentication credential on the IP phone to finish phone provisioning.



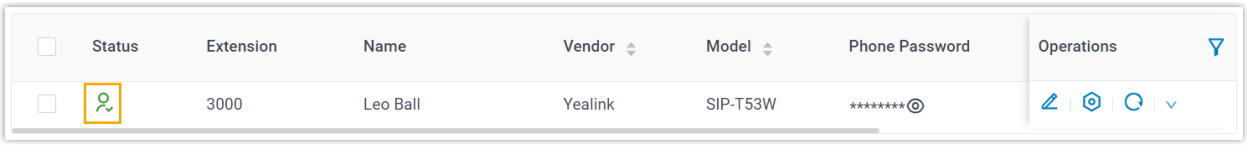
- **Username**: Enter the extension number that is assigned to the phone.
- **Password**: Enter the extension's Voicemail Access PIN.

> **Note:**
> Check the Voicemail Access PIN in the **Voicemail** tab on the extension configuration page.

Result

The extension registration status of provisioned phones is displayed on **Auto Provisioning > Phones**.

- : The assigned extension is registered on the phone.
- : The assigned extension is unregistered on the phone.



Manually Provision an IP Phone

This topic takes the Grandstream GRP2602 IP phone with a firmware version of 1.0.3.67 to describe how to manually provision an IP phone.

> **Note:**
> This topic describes how to provision an IP phone and assign a user's extension to the phone. If you want to set up a hot desking phone via auto provisioning, see Set up a Hot Desking Phone.

Supported IP Phones

This topic can be applied to the Grandstream IP phones listed in Auto Provisioning - Supported Devices.

Prerequisites

- Make sure that you have downloaded the template for the desired phone model (Path: **Auto Provisioning > Resource Repository > Default Templates**).
- Gather information of IP phone, including Vendor, Model, and MAC address.
- RESET the IP phone if it is previously used.

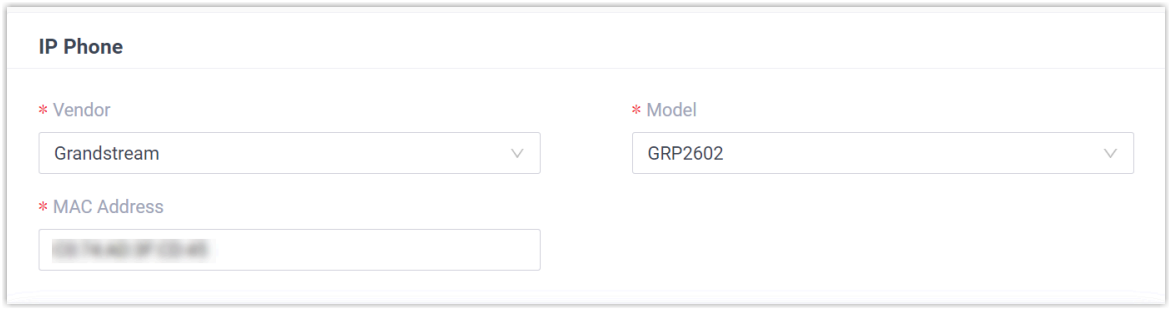
Procedure

Step1. Add phone's MAC address on the PBX

Step2. Configure provisioning server address on the phone

Step1. Add phone's MAC address on the PBX

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Click **Add** to add the IP phone to PBX.
3. In the **IP Phone** section, configure phone information as follows:



   - **Vendor**: Select a phone vendor. In this example, select **Grandstream**.
   - **Model**: Select a phone model. In this example, select **GRP2602**.
   - **MAC Address**: Enter the MAC address of the IP phone.
4. In the **Options** section, configure the auto provision settings.
   - **Template**: Select a desired template from the drop-down list.

     Note:
     The template provides configurations except extension assignment. You can select the default template corresponding to the phone model, or customize your own template. For more information, see Create a Custom Auto Provisioning Template.
   - **Provisioning Link**: A provisioning server URL is generated automatically and displayed on the web page.

     Note:
     Note down the generated provisioning link, as you will use it later.
5. In the **Assign Extension** section, assign an extension to the IP phone.

Tip:
If your desired extension is not listed in the drop-down list, you can check if the extension has been associated with other IP phone.
- To release the previous phone, see Release an Extension from a Provisioned IP Phone.

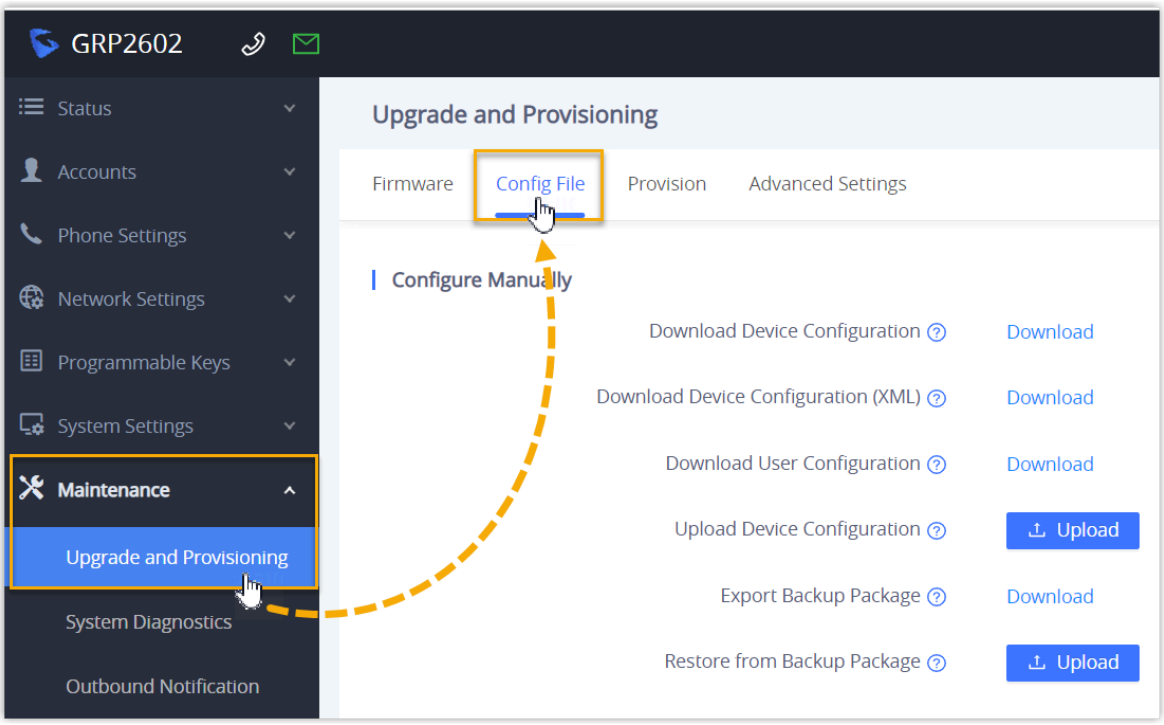
> • To associate an extension with multiple IP phones, see Allow Multiple Registrations for One Extension Number.

6. Click **Save**.

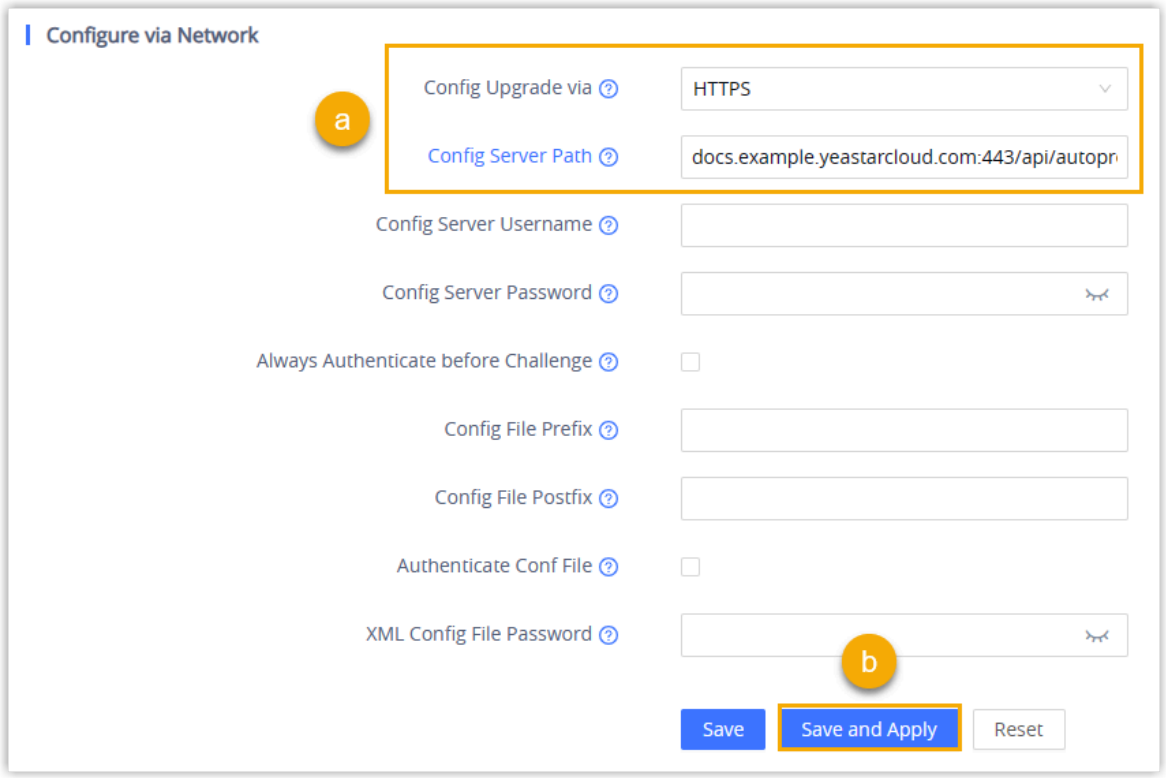
   A configuration file for the IP phone is generated in the PBX.

Step2. Configure provisioning server address on the phone

1. Log in to the IP phone web page by the IP address.
2. On the left navigation bar, go to **Maintenance > Upgrade and Provisioning > Config File**.



3. In the **Configure via Network** section, complete the following settings.



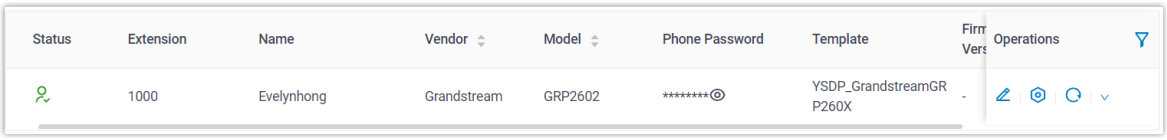
a. Enter the information of the provisioning server.
   - **Config Upgrade via**: Select **HTTPS**.
   - **Config Server Path**: Paste the provisioning link obtained from PBX.

   > **Note:**
   > You should remove the prefix `https://` before pasting the link into the field.

b. Click **Save and Apply**.

Result

- You can check the extension registration status of provisioned phone on PBX (Path: **Auto Provisioning > Phones**).
  - : The assigned extension is registered on the phone.
  - : The assigned extension is unregistered on the phone.



- In future use, the IP phone will automatically download the configuration file from the PBX, and applies the configurations.

Related information

Auto Provisioning - Supported Devices

Manage Provisioned Devices

Reboot Provisioned IP Phones

For some settings that need a device reboot to take effect, you can reboot the device remotely on PBX web portal if these devices have been auto provisioned on PBX.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Reboot phones according to your needs:
   - To reboot a phone, hover your mouse over ⌄ beside the desired phone, and click **Reboot**.
   - To reboot phones in bulk, select the checkboxes of desired phones, and click **Reboot**.

   The system prompts you whether to reboot the phones.
3. Click **OK**.

Reassign an Extension to a Provisioned IP Phone

If a phone is previously provisioned but the phone owner has been changed, you can reassign an extension to the provisioned phone.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**, edit the desired phone.
2. In the **Assign Extension** section, select a desired extension.

> **Tip:**
> If your desired extension is not listed in the drop-down list, you can check if the extension has been associated with other IP phone.
> - To release the previous phone, see Release an Extension from a Provisioned IP Phone.
> - To associate an extension with multiple IP phones, see Allow Multiple Registrations for One Extension Number.

3. Click **Save**.

Result

The extension is automatically registered on the phone, and configurations in the selected template are applied to the phone.

Release an Extension from a Provisioned IP Phone

When an employee resigns or doesn't need the device that is currently bound with the employee's extension, you can release the employee's extension from the device. This topic describes how to release an extension from a provisioned device.

Procedure

1. Release the extension from previous phone.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit a desired extension.
   b. Click the **Phone** tab.
   c. Click **Release From Phone** and **Yes**.
   d. Click **Save**.

   The extension is released from the phone.
2. Reprovision the phone to de-register the extension.

   Go to **Auto Provisioning > Phones**, click beside the phone from which you want to release extension.

Remove IP Phones from Provisioning List

The provisioning list always displays all the devices that are discovered. For the out-of-use devices, you can remove them from the phone provisioning list manually.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Remove phones according to your needs:
   - To remove a phone, hover your mouse over ⌄ beside the desired phone, and click **Delete**.
   - To remove phones in bulk, select the checkboxes of the desired phones, and click **Delete**.
3. Click **OK**.

Result

The system erases all configuration files for the phone and releases the assigned extension.

Export and Import Auto Provisioning Phone Information

The phones' information in the Auto Provisioning phone list can be exported and saved as a template. You can fill in the information of desired phones in the exported file, and import the file to PBX again, so as to add phones to the PBX in bulk conveniently. This topic describes how to export and import Auto Provisioning phone information.

Export Auto Provisioning phone information

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. At the top of the Auto Provisioning phone list, click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Auto Provisioning Phone Information Parameters.

Import Auto Provisioning phone information

We recommend that you export Auto Provisioning phone information to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Learn about the requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Auto Provisioning Phone Information Parameters.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. At the top of the Auto Provisioning phone list, click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The phone information in the CSV file will be displayed in the Auto Provisioning phone list.

Related information

Import and Export -FAQ

Auto Provisioning Options

IP Phone Auto Provisioning Options

Yeastar P-Series Cloud Edition provides a variety of Auto Provisioning options for IP phones that can meet general needs for end users. This topic introduces the settings that can be configured via Auto Provisioning.

General settings (preferences & codecs)

General settings provide the most common needs for extension users, such as phone language, date and time, etc. These settings can be auto provisioned by a template, so that the settings can be applied to multiple devices globally.

For more information, see Apply a New Template to a Provisioned IP Phone.

Extension registration

An extension will be registered on the device after Auto Provisioning. If you change extension registration settings (such as registration password, registration name), you need to re-provision your devices.

> **Note:**
> **Limit of extension registration**
>
> - **For IP phone**: Only one extension can be assigned to a phone via Auto Provisioning.
> - **For DECT phone**: Each handset registers with an extension via Auto Provisioning.

For more information, see Auto Provision IP Phones and Reassign an Extension to a Provisioned IP Phone.

Function key

Various function keys are available for you to customize for each extension user, such as BLF, speed dial, etc. The function keys are associated with extensions, and can be applied when auto provisioning phones.

For more information about the function keys, see Auto Provision Function Keys for Phones.

Device firmware

Yeastar P-Series Cloud Edition allows you to update the device firmware in bulk by Auto Provisioning.

For more information about firmware update, see Update Phone Firmware via Auto Provisioning.

Additional settings

In addition to the above settings, if you need to configure additional settings for the devices, you can also customize a template with additional parameters, and provision devices globally to apply the additional settings.

For more information, see Create a Custom Auto Provisioning Template.

Auto Provision Function Keys for Phones

Each extension user can set his or her own function keys, you can also set up for them. These function keys can be applied to auto-provisioned IP phones, Linkus Desktop Client, Linkus Web Client, and Chrome extension 'Yeastar Linkus for Google'. This topic describes how to provision function keys for extension users' IP phones.

Supported key types

The following table lists the function keys that you can assign for an extension user:

Key type	Function
N/A	No functionality.
Line	Configure line keys for IP phone. **Note:** The key type **Line** is only available for IP phones.
BLF	• Extension ◦ Monitor the call status of a specific extension. ◦ Monitor the DND (Do Not Disturb) presence of a specific extension. ◦ Place a call to the monitored extension. ◦ Pick up calls ringing on the monitored extension. For more information about configuration, see Monitor Extension Status by BLF Key. • Trunk ◦ Monitor the status of a specific trunk. ◦ Place an outbound call through the monitored trunk. For more information about configuration, see Seize a Trunk to Call Out by BLF Key. • Queue ◦ Monitor specific pause status of a queue agent. For more information about configuration, see Monitor Specific Pause Status of an Agent by Function Key.
Speed Dial	Place a call to the most commonly dialed numbers or extensions.
Check Voicemail	• Monitor the status of voicemail. • Check voicemail messages.
Check Group Voicemail	• Monitor the status of group voicemail in shared mode. • Check group voicemail messages.
Park & Retrieve	• Monitor the status of a specific parking number. • Park a call on a specific parking number.

Key type	Function
	• Retrieve a parked call from a specific parking number.
Intercom	Place an intercom call to the monitored extension to make an announcement.
DTMF	Send DTMF signals directly instead of manually entering the numbers each time.
Agent Login/Logout	• Monitor login status in a specific queue. • Log in to or log out of a specific queue.
Agent Pause/Unpause	• Monitor service status in a specific queue. • Pause or unpause receiving a call from a specific queue.
LDAP Directory	Quickly access the LDAP phonebook to query contact information on IP phones. **Note:** The key type **LDAP Directory** is only available for IP phones.
Boss-Secretary Feature	• For boss extensions: Monitor Secretary's Call Status. • For secretary extensions: Monitor Boss's Call Status.
Call Forwarding	Quickly enable or disable call forwarding for the IP phone. **Note:** The key type **Call Forwarding** is only available for IP phones. For more information, see Forward All Incoming Calls to Another Destination by BLF Key.

Supported methods

You can provision function keys for user's IP phone either using an auto provisioning template or by setting them up directly on PBX web portal.

- To provision function keys using a template, see Create an custom Auto Provisioning template and Apply a new template to a provisioned IP phone.
- To provision function keys on PBX web portal, see the instructions below.

Procedure

1. Assign function keys for extension users.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.

- If you want to assign function keys for a specific extension user, click beside the desired extension.
- If you want to assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.

b. Click the **Function Keys** tab.
c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the excess function keys cannot take effect. However, if the user's IP phone is connected with an expansion module, the excess function keys are automatically applied to the expansion module.

- **Type**: Select a key type.
- **Value**: Configure a desired value based on the key type, such as parking number, queue, or extension.
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extensions haven't been associated with phones, see Auto Provision IP Phones to bind phones with the extensions.
3. If the extensions have been associated with phones, reprovision the phones to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. To reprovision a phone, click beside the phone assigned to this extension user.
   c. To reprovision multiple phones, select the checkboxes of the desired phones, click **Reprovision**.

Result

The phone automatically applies the changes. Check the function key status on the phone to see if the changes are applied.

Related information

Auto Provision Yealink Expansion Module with Yeastar P-Series Cloud Edition

Seize a Trunk to Call Out by BLF Key

This topic describes how to configure a BLF key on your IP phone via Auto Provisioning to monitor the PBX trunk, and press the BLF key to quickly place an outbound call through the monitored trunk.

Prerequisites

- A phone is connected to Yeastar P-Series Cloud Edition via Auto Provisioning, and has been assigned with an extension.

  For more information, see Auto Provision IP Phones.

- If you want to seize a trunk to call out by BLF key, make sure the extension assigned to the provisioned phone has the permission to use the monitored trunk for outbound calls.

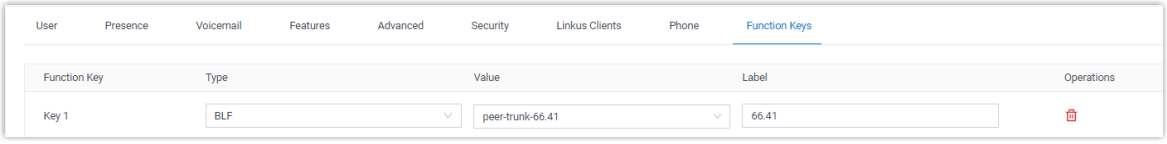
Procedure

- Step1. Set up a function key for trunk monitoring
- Step2. Apply the configuration to IP phone

Step1. Set up a function key for trunk monitoring

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the extension that is assigned to the phone.
2. Click the **Function Keys** tab.
3. Configure a function key to monitor the status of a trunk.

   The following figure shows a configuration example of monitoring the trunk `peer-trunk-66.41`.

   - **Type**: Select **BLF**.
   - **Value**: Enter the name of the trunk to be monitored. In this example, enter `peer-trunk-66.41`.
   - **Label**: Optional. Enter a value, which will be displayed on the phone screen.
4. Click **Save**.

Step2. Apply the configuration to IP phone

1. Go to **Auto Provisioning > Phones**, click beside the desired phone.

   The system prompts you whether to reprovision the phone.
2. In the pop-up window, click **OK**.

Result

- The BLF key shows the real-time status of the monitored trunk:
  - **Green BLF LED**: The trunk is being monitored, and the status is idle.
  - **Red BLF LED**: The trunk is busy.
  - **BLF LED off**: The BLF key configuration failed.
- Press the BLF key to seize the trunk, you will get a dial tone, then dial the number that you want to call.

Forward All Incoming Calls to Another Destination by BLF Key

You can set up call forwarding for provisioned IP phones through Yeastar P-Series Cloud Edition. This topic describes how to set a BLF key on the IP phone via Auto Provisioning, so that users can quickly enable forwarding for all incoming calls by the BLF key.

> **Note:**
> This feature is independent of the extension's call forwarding settings based on presence or feature code. Call forwarding by BLF key is only available for IP phone and will forward all incoming calls to the predefined destination.

Prerequisites

A phone is connected to Yeastar P-Series Cloud Edition via Auto Provisioning, and has been assigned with an extension.

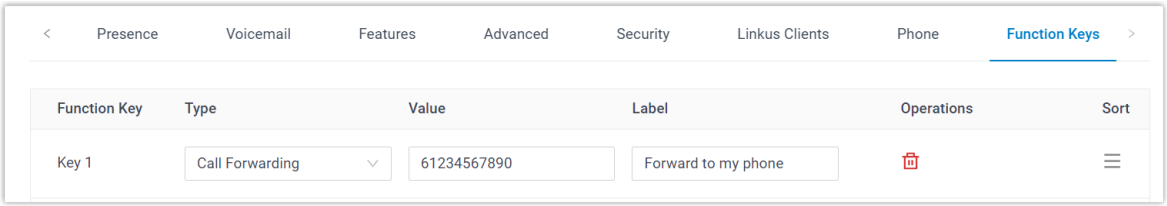
For more information, see Auto Provision IP Phones.

Procedure

- Step 1. Set up a function key for call forwarding
- Step 2. Apply the configuration to IP phone

Step 1. Set up a function key for call forwarding

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the extension that is assigned to the phone.
2. Click the **Function Keys** tab.
3. Configure a function key to set up call forwarding.



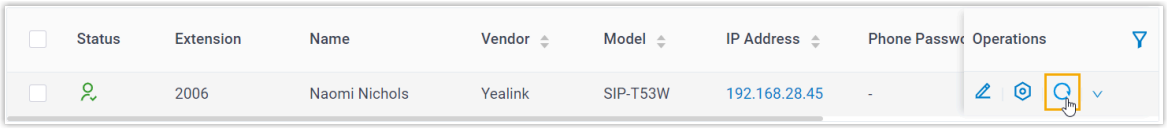
   - **Type**: Select **Call Forwarding**.
   - **Value**: Enter the number to which you want to forward all calls.

     Note:
     If the number is an external number, make sure that the extension associated with the phone has permission to use the outbound route, or the call forwarding would fail.

   - **Label**: Optional. Enter a value, which will be displayed on the phone screen.
4. Click **Save**.

Step 2. Apply the configuration to IP phone

1. On PBX web portal, go to **Auto Provisioning > Phones**, click beside the associated IP phone.



2. In the pop-up window, click **OK**.

Result

- The BLF key shows the current status of the call forwarding:
  - **Green BLF LED**: The call forwarding is enabled.
  - **BLF LED off**: The call forwarding is disabled.

- User can quickly enable the call forwarding by pressing the BLF key, and all incoming calls sent to the extension will be forwarded to the preset destination.

  In this example, all incoming calls sent to extension 2006 will be forwarded to an external number 61234567890.

Modify a Provisioned Phone Settings

Centralized provisioning enables you to configure phones with the same settings, you can also customize settings for a specific phone after provisioning. This topic describes how to modify general settings for an IP phone and a DECT phone.

Modify settings of a provisioned IP phone

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Modify the phone that is associated with the extension.
   a. Click the **Phone** tab.
   b. Modify phone settings in the **Preference** and **Codecs** sections.
   c. Click **Save**.

   > **Note:**
   > If you want to change other settings, click the phone IP address displayed on the provisioning list to access the phone web interface, and change the configurations as your need.
3. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to the extension user.

   The phone automatically applies the changes.

Modify settings of a provisioned DECT phone

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**, edit a desired DECT phone.
2. Modify phone settings in the **Preference** and **Codecs** sections, and click **Save**.
3. On the phone provisioning list, click beside the desired DECT phone to reprovision the phone.

The phone automatically applies the changes.

Manage Auto Provisioning Templates

Apply a New Template to a Provisioned IP Phone

If you want to customize a device, you can create a custom template and apply the new template to the IP phone.

Prerequisites

Create a custom Auto Provisioning template.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**, edit the desired phones.
2. In the **Options** section, select a desired template from the **Template** drop-down list.
3. Click **Save**.

Result

The configurations in the new template will be applied automatically to the phone.

Related information

Update Auto Provisioning template(s) to all applicable devices

View a Default Auto Provisioning Template

The default template of different models contains different parameters, you can view what configurations are included in the default template. This topic describes how to search and view a default template.

Background information

Yeastar P-Series Cloud Edition provides various default templates for each supported device. Devices of different models may share the same template. For example, the template `YSDP_YealinkT5xS` of Yealink applies to Yealink T52S and T54S.

The value of default template

The default template contains general settings that are pre-defined based on device model. There are two types of parameter value in the template: variables and absolute value.

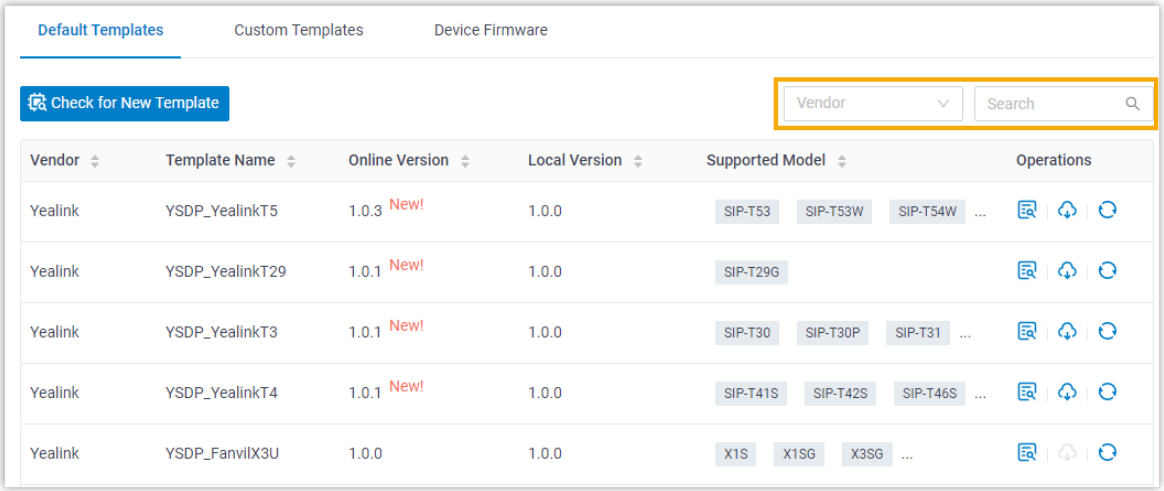
- **Variables**: Variables are attributes to which various values can be assigned. A variable starts with `{{.`, and ends with `}}`. For example, `{{.PhoneWebLanguage}}` means a variable of Phone Web Language setting. The phone web language varies on each phone according to specific phone configuration.
- **Absolute value**: Absolute is a fixed value that applies to all devices that use this template. For example, `features.dtmf.hide_delay = 1` means setting the parameter value to 1 (Enabled).

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository**.
2. Select a device vendor or enter a keyword.

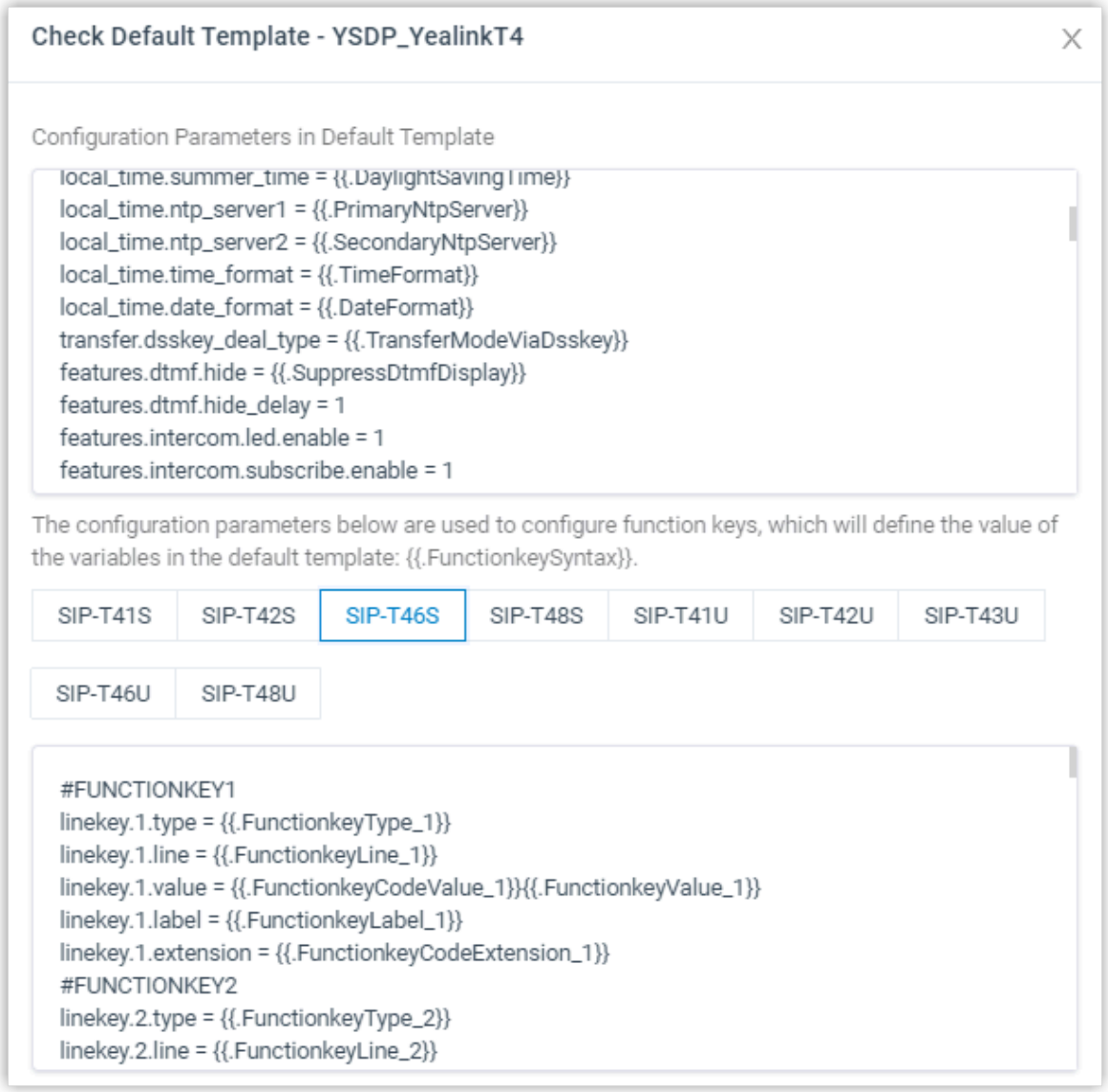
   You can search the template by vendor, provisioning template name, or device model.

   The search results are displayed automatically on the web page.



3. Click beside the desired template to view the default configurations.
   The following figure shows a default template of Yealink T56A. The default template consists of two parts:
   - **Configuration parameters in Default Template**: The pre-defined configuration parameters in this template are displayed in the first text box.
   - **Function keys of device model**: The pre-defined function keys supported by the device model are displayed in the second text box.

You can click the device model tab to view the supported keys.



Related information

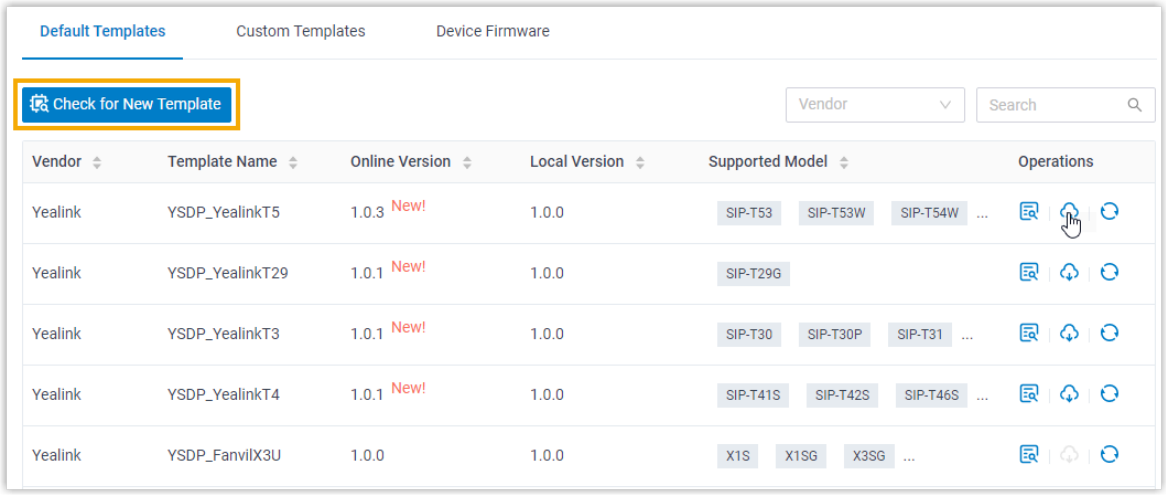
Create a Custom Auto Provisioning Template
Manage Custom Auto Provisioning Templates

Update a Default Auto Provisioning Template

Yeastar P-Series Cloud Edition regularly provides new template versions to release new features and fix bugs. You can check if a new template is available, and decide whether to update the default template. This topic describes how to update a default template.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository**.
2. Click **Check for New Template** to obtain the new template.
3. If a new template is detected, click to download the new template.



4. Click beside the desired template to view the default configurations.

   For more information, see View a Default Auto Provisioning Template.

What to do next

To apply the new template to the devices that have been auto provisioned by the same template with previous version, click .

Create a Custom Auto Provisioning Template

If you want to customize the general settings defined in a default provisioning template, or you want to add custom parameters, you can create a custom Auto Provisioning template. This topic describes how to create a custom Auto Provisioning template.

Background information

Custom template allows you to customize device settings. You can easily modify and apply the custom template to a group of devices, or an individual device.

Yeastar P-Series Cloud Edition provides two types of custom template:

- **Basic Custom Template**: Allow you to customize the parameters provided in the default template.
- **Advanced Custom Template**: Allow you to customize parameters provided in the default template, and add additional parameters for the desired phones.

> **Note:**
> Contact your vendor to make sure that the added parameters are supported.

Create a Auto Provisioning basic template

If you want to customize the settings that are defined in a default provisioning template, you can create a basic Auto Provisioning template.

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Custom Templates**.
2. Click **Add**.
3. In the **Basic** section, set basic information.
   - **Template Name**: Enter a name to help you identify it.
   - **Source Default Template**: Select a default provisioning template to customize.
   - **Template Type**: Select **Basic**.

     The general settings that the source default template provides will be displayed in the **Preference**, **Distinctive Ringtone**, and **Codecs** sections.
   - **Remark**: Optional. Enter a short description about this template.
4. In the **Preference** section, modify the preference settings that are provided by the source default template.
5. In the **Codecs** section, select a desired codec according to your needs.
6. Click **Save**.

Create an advanced Auto Provisioning template

If the settings that you want to configure for your devices are not defined in the default provisioning template, you can create an advanced Auto Provisioning template.

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Custom Templates**.
2. Click **Add**.
3. In the **Basic** section, set the basic information.

- **Template Name**: Enter a name to help you identify it.
- **Source Default Template**: Select a default template to customize.
- **Template Type**: Select **Advanced**.

  The general settings that the source default template provides will be displayed in the **Preference**, **Distinctive Ringtone**, and **Codecs** sections; A text box containing all configuration parameters will be displayed in the **Customize Configuration Parameters In Text** section.
- **Remark**: Optional. Enter a short description about this template.

4. In the **Preference** section, modify the preference settings that are provided by the source default template.
5. In the **Codecs** section, select a desired codec according to your needs.
6. Add additional parameters that are not provided by the source default template.

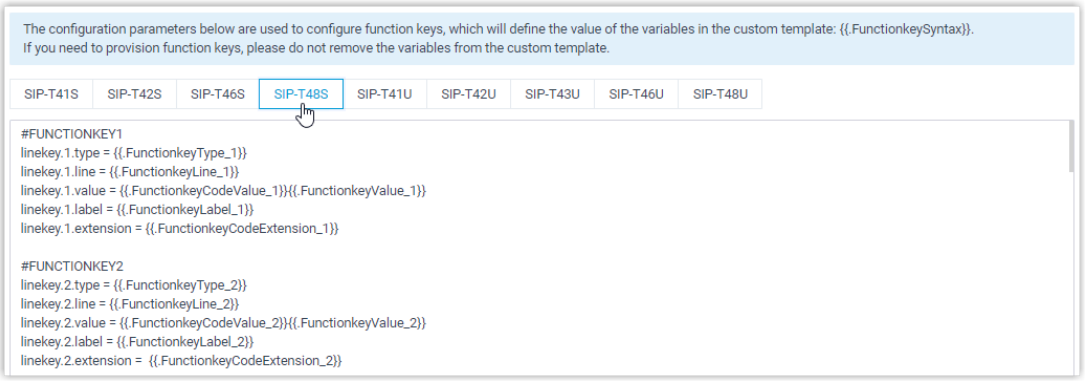
> **Note:**
> The general settings defined in source default template are assigned with variables. The variable that starts with `{{.` and ends with `}}` is associated with the configuration that can be configured on **Preference**, **Codecs**, and **Function Keys** sections. Please don't change the variable if you want to modify the settings from PBX web portal.

   a. In the **Customize Configuration Parameters In Text** section, add your configuration parameters in the first text box.

   > **Note:**
   > Contact your vendor to make sure that the parameters are supported for the device model.

   b. In the second text box, select which function keys to be applied according to the phone model.

   You can also add your function key parameters in the second text box.



7. Click **Save**.

Related information

Edit a custom Auto Provisioning template
Update Auto Provisioning template(s) to all applicable devices

Manage Custom Auto Provisioning Templates

This topic describes how to edit or delete custom Auto Provisioning templates.

Edit a custom Auto Provisioning template

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Custom Templates**.
2. Click beside a desired custom template.
3. Modify the device settings.
4. Click **Save**.
   The system prompts you whether to update the new configurations to devices that use this template.
   - **Yes**: The system generates new configuration files and immediately triggers provisioning for all devices that use this template.
   - **No**: The system saves the changes to this template, and generates new configuration files for all devices that use this template. You can trigger provisioning manually for specific devices later.

Delete custom Auto Provisioning templates

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Custom Templates**.

2. Delete custom templates according to your needs.
    - To delete a custom template, click beside the desired template.
    - To delete custom templates in bulk, select the checkboxes of desired templates, click **Delete**.
3. In the pop-up dialog box, click **Yes**.

    If the template is in use, you need to release it from the devices that use the template first.

Update Auto Provisioning template(s) to all applicable devices

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Custom Templates**.
2. Update the configurations to the devices:
    - To update the configuration of a specific template, click beside the desired template.
    - To update the configuration of multiple templates, select the checkboxes of desired templates, click **Update to Device**.
3. Click **Yes** to trigger phone provisioning.

Manage IP Phone Firmware

Update Phone Firmware via Auto Provisioning

This topic describes how to update phone firmware via Auto Provisioning.

Prerequisites

Upload the desired phone firmware to PBX. For more information, see Add a device firmware.

Update firmware to all applicable phones

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Device Firmware**.
2. Click beside the desired firmware.
3. Click **Yes** to upgrade the phones.

Update firmware to specific phones

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Select the checkboxes of the desired phones.
3. Click **Firmware Upgrade**.
4. Select the firmware that you want to upgrade, click **Upgrade Now**.

Result

The phones automatically reboot and update their firmwares to the new version.

Manage Device Firmware Files

This topic describes how to manage device firmwares, including add, edit, and delete device firmware files.

Add a device firmware file

> **Note:**
> You can upload up to **3** device firmware files to PBX server.

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Device Firmware**, click **Add**.
2. In the **Device** section, select a firmware vendor and device model.
3. In the **Firmware** section, upload the firmware.
   - **Firmware Version**: Enter a name (firmware version) to help you identify it.
   - **Upload Firmware File**: Click **Browse** and select the corresponding firmware.
   - **Remark**: Optional. Enter a short description about the firmware.
4. Click **Save**.

The uploaded firmware is displayed on the **Device Firmware** list. When you update phone firmware, the uploaded firmware can be detected and displayed for you to choose.

Edit a device firmware file

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Device Firmware**.

2. Click beside the desired firmware.
3. In the **Firmware** section, edit the firmware information or update the firmware file.
    - **Firmware Version**: Enter a name (firmware version) to help you identify it.
    - **Upload**: Click **Browse** and select the corresponding firmware.
    - **Remark**: Optional. Edit the note.
4. Click **Save**.

Delete device firmware files

1. Log in to PBX web portal, go to **Auto Provisioning > Resource Repository > Device Firmware**.
2. Delete device firmware files.
    - To delete a device firmware, click beside the desired firmware.
    - To delete device firmwares in bulk, select the checkboxes of the desired firmware, and click **Delete**.
3. Click **OK**.

Auto Provisioning - Supported Devices

This topic lists the devices that are currently supported for Auto Provisioning by Yeastar P-Series Cloud Edition.

Yealink phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Yealink IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Yealink IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
CP920	78.85.0.5 or later	84.5.0.86 or later

Model	Phone Requirement	PBX Requirement
CP925	148.86.0.5 or later	84.5.0.86 or later
CP960	73.85.0.5 or later	84.5.0.86 or later
CP965	143.86.0.5 or later	84.5.0.86 or later
SIP-CP935W	149.86.0.5 or later	84.6.0.24 or later
SIP-T19P_E2	53.84.0.125 or later	84.5.0.86 or later
SIP-T21_E2	52.84.0.125 or later	84.5.0.86 or later
SIP-T21P_E2	52.84.0.125 or later	84.5.0.86 or later
SIP-T23P	44.84.0.125 or later	84.5.0.86 or later
SIP-T23G	44.84.0.125 or later	84.5.0.86 or later
SIP-T27G	69.85.0.5 or later	84.5.0.86 or later
SIP-T29G	46.83.0.120 or later	84.5.0.86 or later
SIP-T30	124.85.0.15 or later	84.5.0.86 or later
SIP-T30P	124.85.0.15 or later	84.5.0.86 or later
SIP-T31	124.85.0.15 or later	84.5.0.86 or later
SIP-T31G	124.85.0.15 or later	84.5.0.86 or later
SIP-T31P	124.85.0.15 or later	84.5.0.86 or later
SIP-T31W	124.86.0.75 or later	84.12.0.32 or later
SIP-T33G	124.85.0.15 or later	84.5.0.86 or later
SIP-T33P	124.85.0.15 or later	84.5.0.86 or later
SIP-T34W	124.86.0.75 or later	84.12.0.32 or later
SIP-T40P	54.84.0.125 or later	84.5.0.86 or later
SIP-T40G	76.84.0.125 or later	84.5.0.86 or later
SIP-T41P	36.83.0.120 or later	84.5.0.86 or later
SIP-T41S	66.85.0.5 or later	84.5.0.86 or later
SIP-T41U	108.85.0.39 or later	84.5.0.86 or later
SIP-T42G	29.83.0.120 or later	84.5.0.86 or later
SIP-T42S	66.85.0.5 or later	84.5.0.86 or later
SIP-T42U	108.85.0.39 or later	84.5.0.86 or later
SIP-T43U	108.85.0.39 or later	84.5.0.86 or later
SIP-T44U	108.86.0.90 or later	84.10.0.32 or later
SIP-T44W	108.86.0.90 or later	84.10.0.32 or later

Model	Phone Requirement	PBX Requirement
SIP-T46G	28.83.0.120 or later	84.5.0.86 or later
SIP-T46S	66.85.0.5 or later	84.5.0.86 or later
SIP-T46U	108.85.0.39 or later	84.5.0.86 or later
SIP-T48G	35.83.0.120 or later	84.5.0.86 or later
SIP-T48S	66.85.0.5 or later	84.5.0.86 or later
SIP-T48U	108.85.0.39 or later	84.5.0.86 or later
SIP-T52S	70.84.0.70 or later	84.5.0.86 or later
SIP-T53	96.85.0.5 or later	84.5.0.86 or later
SIP-T53W	96.85.0.5 or later	84.5.0.86 or later
SIP-T54S	70.84.0.70 or later	84.5.0.86 or later
SIP-T54W	96.85.0.5 or later	84.5.0.86 or later
SIP-T56A	58.83.0.15 or later	84.5.0.86 or later
SIP-T57W	96.85.0.5 or later	84.5.0.86 or later
SIP-T58	58.85.0.5 or later	84.5.0.86 or later
SIP-T58W	150.86.0.5 or later	84.5.0.86 or later
VP59	91.85.0.5 or later	84.5.0.86 or later
W60B (W53P, W41P, W60P, CP930W-Base)	77.83.0.85 or later	84.5.0.86 or later
W70B (W79P, W76P, W73P)	146.85.0.20 or later	84.5.0.86 or later
W75DM	175.85.0.5 or later	84.14.0.26 or later
W80B	W80DM-103.83.0.80	84.5.0.86 or later
W90DM	130.85.0.15 or later	84.5.0.86 or later

Fanvil phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Fanvil IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Fanvil IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
A10	2.12.4 or later	84.11.0.22 or later
A10W	2.12.4 or later	84.11.0.22 or later
A32	2.6.0.408 or later	84.5.0.86 or later
A32i	2.6.0.408 or later	84.5.0.86 or later
A320	2.6.0.1402 or later	84.11.0.22 or later
A320i	2.6.0.1402 or later	84.11.0.22 or later
FH-S01	2.12.8 or later	84.9.0.20 or later
H1	2.12.1 or later	84.10.0.32 or later
H2U	2.4.7 or later	84.5.0.86 or later
H2U-V2	2.4.7.6 or later	84.5.0.86 or later
H3	2.12.1.7334 or later	84.5.0.86 or later
H3W	2.4.4 or later	84.5.0.86 or later
H4	1.0.8 or later	84.14.0.26 or later
H4W	1.0.8 or later	84.14.0.26 or later
H5	2.12.1.7334 or later	84.5.0.86 or later
H5W	2.4.4 or later	84.5.0.86 or later
H6	1.0.8 or later	84.14.0.26 or later
H6W	1.0.8 or later	84.14.0.26 or later
i10	1.2.7 or later	84.5.0.86 or later
i10D	1.2.7 or later	84.5.0.86 or later
i10S	2.4.4 or later	84.5.0.86 or later
i10SD	2.4.4 or later	84.5.0.86 or later
i10SV	2.4.4 or later	84.5.0.86 or later
i10V	1.2.7 or later	84.5.0.86 or later
i11S	1.2.7 or later	84.5.0.86 or later
i11SV	2.4.4 or later	84.5.0.86 or later
i12	2.8.2.7009 or later	84.5.0.86 or later
i16V	2.8.2.7009 or later	84.5.0.86 or later
i16S	2.4.4 or later	84.5.0.86 or later
i16SV	2.4.4 or later	84.5.0.86 or later
i18S	2.8.2.7009 or later	84.5.0.86 or later

Model	Phone Requirement	PBX Requirement
i20S	2.8.2.7009 or later	84.5.0.86 or later
i23S	2.8.2.7009 or later	84.5.0.86 or later
i30	2.8.2.7009 or later	84.5.0.86 or later
i31S	2.8.2.7009 or later	84.5.0.86 or later
i32V	2.8.2.7009 or later	84.5.0.86 or later
i33V	2.8.2.7009 or later	84.5.0.86 or later
i33VF	2.8.2.7009 or later	84.5.0.86 or later
i504	2.12.43.13 or later	84.6.0.24 or later
i505	2.6.6.391 or later	84.11.0.22 or later
i506W	2.12.43.13 or later	84.6.0.24 or later
i507W	2.6.6.394 or later	84.11.0.22 or later
i51	2.8.13 or later	84.5.0.86 or later
i51W	2.8.13 or later	84.5.0.86 or later
i52	2.8.13 or later	84.5.0.86 or later
i52W	2.8.13 or later	84.5.0.86 or later
i53	2.8.13 or later	84.5.0.86 or later
i53W	2.8.13 or later	84.5.0.86 or later
i55A	1.0.0.45 or later	84.8.0.25 or later
i56A	0.3.0.21 or later	84.5.0.86 or later
i57A	1.0.0.46 or later	84.8.0.25 or later
i61	2.4.0 or later	84.6.0.24 or later
i62	2.4.0 or later	84.6.0.24 or later
i63	2.4.0 or later	84.6.0.24 or later
i64	2.4.0 or later	84.6.0.24 or later
i68	2.8.40.22 or later	84.8.0.25 or later
PA2	2.8.2.7009 or later	84.5.0.86 or later
PA2S	2.8.11 or later	84.5.0.86 or later
PA3	2.4.4 or later	84.5.0.86 or later
V61G	2.12.18.8 or later	84.14.0.26 or later
V61W	2.12.18.8 or later	84.14.0.26 or later
V62	2.4.10 or later	84.6.0.24 or later

Model	Phone Requirement	PBX Requirement
V62G	2.12.18.8 or later	84.14.0.26 or later
V62W	2.12.18.8 or later	84.14.0.26 or later
V63	2.12.16.19 or later	84.11.0.22 or later
V64	2.4.10 or later	84.6.0.24 or later
V65	2.12.2.4 or later	84.7.0.17 or later
V67	2.6.0 or later	84.6.0.24 or later
W610W	2.12.0 or later	84.11.0.22 or later
W611W	pvt-2.8 or later	84.8.0.25 or later
X1S / X1SP	2.2.12 or later	84.5.0.86 or later
X1SG	2.2.12 or later	84.5.0.86 or later
X2/X2P	2.14.0.7386 or later	84.5.0.86 or later
X2C/X2CP	2.14.0.7386 or later	84.5.0.86 or later
X210	2.2.11 or later	84.5.0.86 or later
X210-V2	2.12.1.3 or later	84.7.0.17 or later
X210i	2.2.11 or later	84.5.0.86 or later
X210i-V2	2.12.1.3 or later	84.7.0.17 or later
X3SG	2.2.12 or later	84.5.0.86 or later
X3S/X3SP/X3G	2.14.0.7386 or later	84.5.0.86 or later
X3S Lite / X3SP Lite	2.4.5 or later	84.5.0.86 or later
X3S Pro / X3SP Pro	2.4.5 or later	84.5.0.86 or later
X3SW	2.4.5 or later	84.5.0.86 or later
X3SG Lite	2.4.5 or later	84.5.0.86 or later
X3SG Pro	2.4.5 or later	84.5.0.86 or later
X3U	2.2.12 or later	84.5.0.86 or later
X3U Pro	2.4.5 or later	84.5.0.86 or later
X301	0.0.16 or later	84.8.0.25 or later
X301G	0.0.16 or later	84.8.0.25 or later
X301W	0.0.16 or later	84.8.0.25 or later
X303	0.0.16 or later	84.8.0.25 or later
X303G	0.0.16 or later	84.8.0.25 or later
X303W	0.0.16 or later	84.8.0.25 or later

Model	Phone Requirement	PBX Requirement
X303-2 WIRE	1.0.3 or later	84.14.0.26 or later
X305	2.12.1.6 or later	84.8.0.25 or later
X4/X4G	2.14.0.7386 or later	84.5.0.86 or later
X4U	2.2.11 or later	84.5.0.86 or later
X4U-V2	2.12.1 or later	84.6.0.24 or later
X5U	2.2.11 or later	84.5.0.86 or later
X5U-V2	2.12.1 or later	84.6.0.24 or later
X5S	2.2.1 or later	84.5.0.86 or later
X6	2.2.1 or later	84.5.0.86 or later
X6U	2.2.11 or later	84.5.0.86 or later
X6U-V2	2.12.1 or later	84.6.0.24 or later
X7	2.2.11 or later	84.5.0.86 or later
X7A	2.2.0.229 or later	84.5.0.86 or later
X7C	2.2.11 or later	84.5.0.86 or later
X7-V2	2.12.1.3 or later	84.7.0.17 or later
X7C-V2	2.12.1.3 or later	84.7.0.17 or later
Y501	2.12.4 or later	84.11.0.22 or later
Y501W	2.12.4 or later	84.11.0.22 or later
Y501-Y	2.12.4 or later	84.11.0.22 or later
Y501W-Y	2.12.4 or later	84.11.0.22 or later

Grandstream phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Grandstream IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Grandstream IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
GXP1610	1.0.7.13 or later	84.9.0.18 or later
GXP1620	1.0.7.13 or later	84.9.0.18 or later
GXP1625	1.0.7.13 or later	84.9.0.18 or later
GXP1628	1.0.7.13 or later	84.9.0.18 or later
GXP1630	1.0.7.13 or later	84.9.0.18 or later
GXP2130	1.0.11.16 or later	84.9.0.18 or later
GXP2135	1.0.11.16 or later	84.9.0.18 or later
GXP2140	1.0.11.16 or later	84.9.0.18 or later
GXP2160	1.0.11.16 or later	84.9.0.18 or later
GXP2170	1.0.11.16 or later	84.9.0.18 or later
GAC2500	1.0.3.45 or later	84.11.0.22 or later
GAC2570	1.0.1.36 or later	84.11.0.22 or later
GRP2601	1.0.3.63 or later	84.9.0.18 or later
GRP2601P	1.0.3.63 or later	84.9.0.18 or later
GRP2602	1.0.3.63 or later	84.9.0.18 or later
GRP2602P	1.0.3.63 or later	84.9.0.18 or later
GRP2602G	1.0.3.63 or later	84.9.0.18 or later
GRP2602W	1.0.3.63 or later	84.9.0.18 or later
GRP2603	1.0.3.63 or later	84.9.0.18 or later
GRP2603P	1.0.3.63 or later	84.9.0.18 or later
GRP2604	1.0.3.63 or later	84.9.0.18 or later
GRP2604P	1.0.3.63 or later	84.9.0.18 or later
GRP2612	1.0.7.25 or later	84.9.0.18 or later
GRP2612P	1.0.7.25 or later	84.9.0.18 or later
GRP2612G	1.0.7.25 or later	84.9.0.18 or later
GRP2612W	1.0.7.25 or later	84.9.0.18 or later
GRP2613	1.0.7.25 or later	84.9.0.18 or later
GRP2614	1.0.7.25 or later	84.9.0.18 or later
GRP2615	1.0.7.25 or later	84.9.0.18 or later
GRP2616	1.0.7.25 or later	84.9.0.18 or later
GRP2624	1.0.7.25 or later	84.9.0.18 or later

Model	Phone Requirement	PBX Requirement
GRP2634	1.0.7.25 or later	84.9.0.18 or later
GRP2670	1.0.7.25 or later	84.9.0.18 or later

Htek phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Htek IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Htek IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
UC902	2.0.4.8.18 or later	84.5.0.86 or later
UC902S	2.0.4.8.18 or later	84.5.0.86 or later
UC903	2.0.4.8.18 or later	84.5.0.86 or later
UC912	2.0.4.8.18 or later	84.5.0.86 or later
UC912G	2.0.4.8.18 or later	84.5.0.86 or later
UC912E	2.0.4.8.18 or later	84.5.0.86 or later
UC921	2.0.4.8.18 or later	84.5.0.86 or later
UC921G	2.0.4.8.18 or later	84.5.0.86 or later
UC923	2.0.4.8.18 or later	84.5.0.86 or later
UC923U	2.0.4.8.18 or later	84.5.0.86 or later
UC924	2.0.4.8.18 or later	84.5.0.86 or later
UC924E	2.0.4.8.18 or later	84.5.0.86 or later
UC924U	2.0.4.8.18 or later	84.5.0.86 or later
UC924W	2.0.4.8.18 or later	84.5.0.86 or later
UC926	2.0.4.8.18 or later	84.5.0.86 or later
UC926E	2.0.4.8.18 or later	84.5.0.86 or later
UC926U	2.0.4.8.18 or later	84.5.0.86 or later
UCV10	5.42.1.6.30b58 or later	84.12.0.32 or later

Model	Phone Requirement	PBX Requirement
UCV20	5.42.1.6.30b79 or later	84.12.0.32 or later
UCV50	5.42.1.6.30b62 or later	84.12.0.32 or later
UCV52	5.42.1.6.30b68 or later	84.12.0.32 or later
UCV53	5.42.1.6.32R76 or later	84.12.0.32 or later

Gigaset phones

Note:
For more information about the auto provisioning configuration, see Auto Provision Gigaset DECT System with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
N870 IP PRO	2.38.1 or later	84.5.0.86 or later
N870 VI PRO	2.38.1 or later	84.5.0.86 or later
N670 IP PRO	2.38.1 or later	84.5.0.86 or later
N610 IP PRO	2.52.0 or later	84.5.0.86 or later
Maxwell Basic PRO	3.18.1 or later	84.5.0.86 or later
Maxwell 2 PRO	3.18.1 or later	84.5.0.86 or later
Maxwell 3 PRO	3.18.1 or later	84.5.0.86 or later
Maxwell 4 PRO	3.18.1 or later	84.5.0.86 or later

Snom phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Snom IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Snom IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
D120	10.1.54.13 or later	84.5.0.86 or later

Model	Phone Requirement	PBX Requirement
D140	10.1.148.1 or later	84.12.0.34 or later
D150	10.1.148.1 or later	84.12.0.34 or later
D315	10.1.73.16 or later	84.5.0.86 or later
D335	10.1.73.16 or later	84.5.0.86 or later
D385	10.1.73.16 or later	84.5.0.86 or later
D713	10.1.73.16 or later	84.6.0.46 or later
D717	10.1.73.16 or later	84.5.0.86 or later
D735	10.1.73.16 or later	84.5.0.86 or later
D785	10.1.73.16 or later	84.5.0.86 or later
D862	10.1.137.15 or later	84.9.0.22 or later
D865	10.1.137.15 or later	84.9.0.22 or later
HD100	1.0.0.3-0 or later	84.14.0.26 or later
HD101	1.0.0.3-0 or later	84.14.0.26 or later
HD350W	1.0.0.3-0 or later	84.14.0.26 or later
HD351W	1.0.0.3-0 or later	84.14.0.26 or later
HM201	1.0.0.3-0 or later	84.14.0.26 or later
M100 KLE	1.0.5.7 or later	84.14.0.24 or later
M500	1.12.2 or later	84.14.0.24 or later
M300	BSV530B2 or later	84.8.0.25 or later
M400	BSV610B5 or later	84.8.0.25 or later
M900	BSV530B7 or later	84.8.0.25 or later

Flyingvoice phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Flyingvoice IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Flyingvoice IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
FIP10	0.7.23.1 or later	84.8.0.25 or later
FIP11C	0.7.23.1 or later	84.8.0.25 or later
FIP12WP	0.7.23.1 or later	84.8.0.25 or later
FIP13G	0.7.23.1 or later	84.8.0.25 or later
FIP14G	0.7.23.1 or later	84.8.0.25 or later
FIP15G	0.7.23.1 or later	84.8.0.25 or later
FIP15G Plus	0.7.23.1 or later	84.8.0.25 or later
FIP16	0.7.23.1 or later	84.8.0.25 or later
FIP16 Plus	0.7.23.1 or later	84.8.0.25 or later
P10	V0.7.56 or later	84.9.0.20 or later
P10P	V0.7.56 or later	84.9.0.20 or later
P10G	V0.7.56 or later	84.9.0.20 or later
P10W	V0.7.56 or later	84.9.0.20 or later
P10LTE	V0.7.56 or later	84.9.0.20 or later
P11	V0.7.56 or later	84.9.0.20 or later
P11P	V0.7.56 or later	84.9.0.20 or later
P11G	V0.7.56 or later	84.9.0.20 or later
P11W	V0.7.56 or later	84.9.0.20 or later
P11LTE	V0.7.56 or later	84.9.0.20 or later
P20	V0.7.57 or later	84.9.0.20 or later
P20P	V0.7.57 or later	84.9.0.20 or later
P20W	V0.7.57 or later	84.9.0.20 or later
P20G	V0.7.57 or later	84.9.0.20 or later
P21	V0.7.57 or later	84.9.0.20 or later
P21P	V0.7.57 or later	84.9.0.20 or later
P21W	V0.7.57 or later	84.9.0.20 or later
flyphone	V0.7.57 or later	84.9.0.20 or later
P22P	V0.7.57 or later	84.9.0.20 or later
P22G	V0.7.57 or later	84.9.0.20 or later
P23G	V0.7.57 or later	84.9.0.20 or later
P23GW	V0.7.57 or later	84.9.0.20 or later

Model	Phone Requirement	PBX Requirement
P24G	V0.7.57 or later	84.9.0.20 or later
i86Box_Basic	V0.0.16.1 or later	84.9.0.20 or later
i86Box_Indoor	V0.0.16.1 or later	84.9.0.20 or later
i86Box_2Line	V0.0.16.1 or later	84.9.0.20 or later
i86Box_PCBA	V0.0.16.1 or later	84.9.0.20 or later
i86Box_NFC	V0.0.16.1 or later	84.9.0.20 or later

Tiptel phones

Note:

- For more information about the auto provisioning configuration, see Auto Provision Tiptel IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Tiptel IP Phone with Yeastar P-Series Cloud Edition.

Table 10.

Model	Phone Requirement	PBX Requirement
3310	2.42.6.5.55 or later	84.7.0.17 or later
3320	2.42.6.5.55 or later	84.7.0.17 or later
3330	2.42.6.5.55 or later	84.7.0.17 or later
3340	2.42.6.5.55 or later	84.7.0.17 or later

Alcatel-Lucent Enterprise phones

Note:

- For more information about the autp provisioning configuration, see Provision Alcatel Lucent Enterprise (ALE) IP Phone with Yeastar P-Series Cloud Edition.

- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Alcatel Lucent Enterprise (ALE) IP Phone with Yeastar P-Series Cloud Edition.

Table 11.

Model	Phone Requirement	PBX Requirement
H2	2.10.00.0001083 or later	84.9.0.18 or later
H2P	2.10.00.0001083 or later	84.9.0.18 or later
H3P	2.12.43.010.2272 or later	84.9.0.18 or later
H3G	2.12.43.010.2272 or later	84.9.0.18 or later
H6	2.12.43.010.2272 or later	84.9.0.18 or later
M3	2.13.37.000.2202 or later	84.9.0.18 or later
M5	2.13.37.000.2202 or later	84.9.0.18 or later
M7	2.13.37.000.2202 or later	84.9.0.18 or later
M8	2.13.32.000.1535 or later	84.9.0.18 or later

Mitel phones

Note:

- For more information about the auto provisioning configuration, see Provision Mitel IP Phone with Yeastar P-Series Cloud Edition.
- For phone models that are not in the list, you can manually register the IP phone with PBX. For more information, see Manually Register Mitel IP Phone with Yeastar P-Series Cloud Edition.

Table 12.

Model	Phone Requirement	PBX Requirement
6863i	R5.1.0SP6 or later	84.11.0.22 or later
6865i	R5.1.0SP6 or later	84.11.0.22 or later
6867i	R5.1.0SP6 or later	84.11.0.22 or later
6869i	R5.1.0SP6 or later	84.11.0.22 or later

Table 12. (continued)

Model	Phone Requirement	PBX Requirement
6873i	R5.1.0SP6 or later	84.11.0.22 or later
6920	6.3.1 SP1 or later	84.11.0.22 or later
6930	6.3.1 SP1 or later	84.11.0.22 or later
6940	6.3.1 SP1 or later	84.11.0.22 or later

Poly phones

Note:
For more information about the auto provisioning configuration, see Auto Provision Poly IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
Edge_E100	8.0.0.15602 or later	84.15.0.22 or later
Edge_E220	8.0.0.15602 or later	84.15.0.22 or later
Edge_E300	8.0.0.15602 or later	84.15.0.22 or later
Edge_E320	8.0.0.15602 or later	84.15.0.22 or later
Edge_E350	8.0.0.15602 or later	84.15.0.22 or later
Edge_E400	8.0.0.15602 or later	84.15.0.22 or later
Edge_E450	8.0.0.15602 or later	84.15.0.22 or later
Edge_E500	8.0.0.15602 or later	84.15.0.22 or later
Edge_E550	8.0.0.15602 or later	84.15.0.22 or later
VVX_101	6.4.3.5059 or later	84.15.0.22 or later
VVX_201	6.4.3.5059 or later	84.15.0.22 or later
VVX_301	6.4.3.5059 or later	84.15.0.22 or later
VVX_310	5.9.8 or later	84.15.0.22 or later
VVX_311	6.4.3.5059 or later	84.15.0.22 or later
VVX_401	6.4.3.5059 or later	84.15.0.22 or later
VVX_410	5.9.8 or later	84.15.0.22 or later
VVX_411	6.4.3.5059 or later	84.15.0.22 or later
VVX_501	6.4.3.5059 or later	84.15.0.22 or later
VVX_601	6.4.3.5059 or later	84.15.0.22 or later

Model	Phone Requirement	PBX Requirement
VVX_150	6.4.3.5059 or later	84.15.0.22 or later
VVX_250	6.4.3.5059 or later	84.15.0.22 or later
VVX_350	6.4.3.5059 or later	84.15.0.22 or later
VVX_450	6.4.3.5059 or later	84.15.0.22 or later

Wildix phones

Note:
For more information about the auto provisioning configuration, see Provision Wildix IP Phone with Yeastar P-Series Cloud Edition.

Model	Phone Requirement	PBX Requirement
WP410R2	50.145.6.169 or later	84.15.0.22 or later
WP480R2	55.145.6.111 or later	84.15.0.22 or later
WP480R3	63.145.10.168 or later	84.15.0.22 or later
WP480R4	65.145.6.38 or later	84.15.0.22 or later
WP490R2	59.145.6.148 or later	84.15.0.22 or later
WP490R3	67.145.8.107 or later	84.15.0.22 or later

Auto Provisioning - Variables in Templates

The provision templates make use of a set of variables that are replaced by the actual value when a device is provisioned. This topic shows you the variables used in the provisioning templates.

Variable	Description
Preference settings	
{{.PhoneWebLanguage}}	The language configured on phone web interface.
{{.PhoneLanguage}}	The language configured on phone interface.
{{.Tones}}	The default ringtone of the phone.
{{.CallWaiting}}	Enable or disable call waiting feature.
{{.PhoneUser}}	The user name for logging in to the phone web interface.

Variable	Description
{{.PhonePassword}}	The password for logging in to the phone web interface.
{{.TimeZone}}	The time zone.
{{.TimeZoneName}}	The time zone name.
{{.DaylightSavingTime}}	The daylight saving time.
{{.PrimaryNtpServer}}	The primary NTP server address.
{{.SecondaryNtpServer}}	The second NTP server address.
{{.TimeFormat}}	The time format.
{{.DateFormat}}	The date format.
{{.DateSeparatorFormat}}	The date separator format.
{{.TransferModeViaDsskey}}	The transfer mode for function key.
{{.SuppressDtmfDisplay}}	Enable or disable the IP phone to suppress the display of DTMF digits.
{{.AutoProvisionServerUrl}}	The URL of provision server.
{{.AutoProvisionServerUrlWithoutProtocol}}	The URL of provision server without transport protocol.
{{.AutoProvisionServerProtocolUppercase}}	The uppercase representation of the transport protocol in the auto provisioning URL.
{{.AutoProvisionServerProtocolLowercase}}	The lowercase representation of the transport protocol in the auto provisioning URL.
{{.ProvisioningFile}}	The name of configuration file.
{{.FirmwareUrl}}	The URL of firmware.
{{.FirmwareUrlWithoutProtocol}}	The URL of firmware without transport protocol.
{{.FirmwareFile}}	The name of firmware.
{{.FirmwareFullDownloadUrl}}	The full download URL of firmware.
{{.FirmwareVersion}}	The version of firmware.
{{.EnableUacsta}}	Enable or disable uaCSTA.
{{.AlertInfoText_X}}	The alert info text to trigger the IP phone to play a specific ring tone.
{{.AlertInfoRingtone_X}}	The specific ring tone corresponding to Alert info.
Contact settings for Yealink / Flyingvoice / Wildix/ Dinstar phones	
{{.CompanyPbUrl}}	The URL of company contact file.
{{.CompanyPbName}}	The name of company contact.

Variable	Description
{{.PersonalPbUrl}}	The URL of personal contact file.
{{.PersonalPbName}}	The name of personal contact.
Account settings for IP phones	
{{.EnbAccount}}	Enable or disable extension registration.
{{.AccountLabel}}	The extension label.
{{.AccountDisplayName}}	The display name of extension.
{{.AccountRegistrationName}}	The registration name of extension.
{{.AccountRegistrationExtNumber}}	The registration number of extension.
{{.AccountRegistrationPassword}}	The registration password of extension.
{{.AccountSipServerAddr}}	The URL of PBX server for extension registration.
{{.AccountSipServerPort}}	The port of PBX server for extension registration.
{{.AccountSipServerTransportType}}	The type of transport protocol for extension registration.
{{.AutoAnswer}}	Enable or disable auto answer feature.
{{.CheckVoicemail}}	The voicemail feature code.
Account settings for DECT phones(x is the handset ID)	
{{.EnbAccount_x}}	Enable or disable extension registration.
{{.SRTP_x}}	Enable or disable extension SRTP feature.
{{.AccountLabel_x}}	The extension label.
{{.AccountDisplayName_x}}	The display name of extension.
{{.AccountRegistrationName_x}}	The registration name of extension.
{{.AccountRegistrationExtNumber_x}}	The registration number of extension.
{{.AccountRegistrationPassword_x}}	The registration password of extension.
{{.AccountSipServerAddr_x}}	The URL of provisioning server for extension registration.
{{.AccountSipServerPort_x}}	The port of provisioning server for extension registration.
{{.AccountSipServerTransportType_x}}	The type of transport protocol for extension registration.
{{.CheckVoicemail_x}}	The voicemail feature code.
SIP server template for Yealink W80B(x is the template ID, X=1, 2 or 3)	
{{.TemplateNamex}}	The template name.
{{.TemplateServerAddrx}}	The URL of PBX server for extension registration.
{{.TemplateServerPortx}}	The port of PBX server for extension registration.
{{.AcountSipServerTemplate}}	The type of transport protocol for extension registration.

Variable	Description
Audio codec(x is the codec priority, X=1, 2, 3 or 4)	
{{.AccountAudioCodec_X}}	The priority of the audio codec.
{{.AudioCodecsPriorities}}	The priority of the audio codec.
{{.AccountCodecPcmu}}	Enable or disable PCMU audio codec.
{{.AccountCodecPcmu_Priority}}	The priority of the PCMU audio codec.
{{.AccountCodecPcma}}	Enable or disable PCMA audio codec.
{{.AccountCodecPcma_Priority}}	The priority of the PCMA audio codec.
{{.AccountCodecIlbc}}	Enable or disable iLBC audio codec.
{{.AccountCodecIlbc_Priority}}	The priority of the iLBC audio codec.
{{.AccountCodecIlbc_15_2_Kbps}}	Enable or disable iLBC_15_2 audio codec.
{{.AccountCodecIlbc_15_2_Kbps_Priority}}	The priority of the iLBC_15_2 audio codec.
{{.AccountCodecIlbc_13_33_Kbps}}	Enable or disable iLBC_13_33 audio codec.
{{.AccountCodecIlbc_13_33_Kbps_Priority}}	The priority of the iLBC_13_33 audio codec.
{{.AccountCodecG722}}	Enable or disable G722 audio codec.
{{.AccountCodecG722_Priority}}	The priority of the G722 audio codec.
{{.AccountCodecG729}}	Enable or disable G729 audio codec.
{{.AccountCodecG729_Priority}}	The priority of the G729 audio codec.
{{.AccountCodecG726_32}}	Enable or disable G726_32 audio codec.
{{.AccountCodecG726_32_Priority}}	The priority of the G726_32 audio codec.
{{.AccountCodecSpeex}}	Enable or disable Speex audio codec.
{{.AccountCodecSpeex_Priority}}	The priority of the Speex audio codec.
{{.AccountAdpcmCodec}}	Enable or disable Adpcm audio codec.
{{.AccountCodecAdpcm_Priority}}	The priority of the Adpcm audio codec.
{{.AccountCodecMpeg4}}	Enable or disable Mpeg4 audio codec.
{{.AccountCodecMpeg4_Priority}}	The priority of the Mpeg4 audio codec.
{{.AccountCodecGsm}}	Enable or disable GSM audio codec.
{{.AccountCodecGsm_Priority}}	The priority of the GSM audio codec.
{{.AccountCodecOpus}}	Enable or disable Opus audio codec.
{{.AccountCodecOpus_Priority}}	The priority of the Opus audio codec.

Variable	Description
Video codec(x is the codec priority, X=1, 2, 3 or 4)	
{{.AccountVideoCodec_X}}	The priority of the video codec.
{{.AccountCodecH264}}	Enable or disable H264 codec.
{{.AccountCodecH264_Priority}}	The priority of the H264 codec.
{{.AccountCodecH264_Hp}}	Enable or disable H264_Hp codec.
{{.AccountCodecH264_Hp_Priority}}	The priority of the H264_Hp codec.
{{.AccountCodecH263}}	Enable or disable H263 codec.
{{.AccountCodecH263_Priority}}	The priority of the H263 codec.
{{.AccountCodecH263_P}}	Enable or disable H263_P codec.
{{.AccountCodecH263_P_Priority}}	The priority of the H263_P codec.
{{.AccountCodecVp8}}	Enable or disable Vp8 codec.
{{.AccountCodecVp8_Priority}}	The priority of the Vp8 codec.
Function key (x is the function key ID)	
{{.FunctionkeyType_x}}	The type of function key.
{{.FunctionkeyType2_x}}	The type of function key (for Dynamic VPK).
{{.FunctionkeySubtype_x}}	The subtype of function key.
{{.FunctionkeyLine_x}}	The extension to which function key applies.
{{.FunctionkeyCodeValue_x}}	The feature code of function key.
{{.FunctionkeyValue_x}}	The object of function key.
{{.FunctionkeyExtension_x}}	The number where the call can be picked up by function key.
{{.FunctionkeyCodeExtension_x}}	The pickup code applied for function key.
{{.FunctionkeyLabel_x}}	The label of function key that is displayed on phone screen.
LDAP Directory (x is the template ID, X=1, 2 or 3)	
{{.EnableLdap_X}}	Enable LDAP directory.
{{.LdapName_X}}	Specify the name of LDAP directory.
{{.LdapMode_X}}	Set up the LDAP mode.
{{.LdapHost_X}}	The address of LDAP Server.
{{.LdapNameFilter_X}}	The LDAP name filter.
{{.LdapNumFilter_X}}	The LDAP number filter.

Variable	Description
{{.LdapNameAttr_X}}	The name attribute returned by LDAP Server.
{{.LdapNumAttr_X}}	The number attribute returned by LDAP Server.
{{.LdapDisplayName_X}}	The name of the search results displayed on IP phones.
{{.LdapMaxHit_X}}	The maximum number of search results to be returned by LDAP Server.
{{.LdapIncomingLookup_X}}	Enable or disable IP phone to perform an LDAP search when receiving an incoming call.
{{.LdapDialLookup_X}}	Enable or disable IP phone to perform an LDAP search when placing a call.
{{.LdapSort_X}}	Enable or disable IP phone to sort out search results in alphabetical and numerical order.
{{.LdapPort_X}}	The port of LDAP Server.
{{.LdapBase_X}}	The base entry of the LDAP directory.
{{.LdapUser_X}}	The user accessing the LDAP Server.
{{.LdapPassword_X}}	The password for accessing to the LDAP Server.
Others	
{{.MacAddress}}	The MAC address of phone. **Note:** Here the value does not need a separator of `:`. For example, `09139876900e`.

User Role

User Roles and Permissions

Yeastar P-Series Cloud Edition allows super administrator to have a role-based control over the PBX features that are accessible and manageable on users' web portals. This topic describes what is a user role, and introduces the pre-defined user roles and their permissions.

What is a user role

A user role includes a set of permissions, which allows super administrator to control what PBX features users can manage on users' web portals.

Super administrator can assign user roles to employees based on their job duties, each user role has different permissions. For example, you can assign **Operator** to an employee who is responsible for security of PBX server and network; assign **Human Resource** to an employee who is responsible for dealing with employee profiles.

Pre-defined user roles

Yeastar P-Series Cloud Edition has pre-defined user roles that cover the most common permission configurations. The pre-defined user roles and their permissions are as follows:

Table 13.

Role	Permission
Super administrator	Access and manage all the PBX features. **Note:** The username of super administrator is created when you first configure the system, and the username is unchangeable.
Administrator	Access and manage all the PBX features except the followings: • View **Dashboard** • Manage **Role**
Supervisor	No access to PBX features.
Operator	Access and manage all the features under **Security** and **Maintenance** modules. For more information, see Security and Maintenance.

Table 13. (continued)

Role	Permission
Employee	No access to PBX features.
Human Resource	View and manage all the extensions.
Accounting	Access and manage **Plan**.

Related information

Create a User Role
Assign a Role to a User

Create a User Role

If the pre-defined roles can not meet your need, you can create a user role and grant permissions to the role. This topic describes how to create a user role.

Restrictions

Note:
Only system super administrator can create a user role.

Create a new role

Based on an employee's job duty, you can create a user role and grant corresponding permissions.

1. Log in to PBX web portal, go to **Extension and Trunk > Role**.
2. Click **Add** to create a new role.
3. In the **Role Name** field, enter a name to help you identify it.
4. Grant permissions to the user role.

   For permission details, see User Role Permissions.
5. Click **Save**.

Create a role by copying an existing role

You can create a role based on an existing user role, the new role automatically inherits permissions from the existing role. After copying permissions, you can add or remove permissions as needed.

1. Log in to PBX web portal, go to **Extension and Trunk > Role**.
2. Create a role.
    a. Click **Copy Role**.
    b. In the **Choose a role to copy** drop-down list, select a role.
    c. In the **Role Name** field, enter a name to help you identify the role.
    d. Click **Save**.

    The new role inherits permissions from the existing role.
3. Update permissions for the newly created role.
    a. On **Role** list, click beside the role that you have created.
    b. Select or unselect the checkboxes of the desired permissions.

    For permission details, see User Role Permissions.
    c. Click **Save**.

What to do next

Assign a Role to a User.

Assign a Role to a User

This topic describes how to assign a role to a user.

Restrictions

> **Note:**
> Only system super administrator can assign a role to a user.

Prerequisites

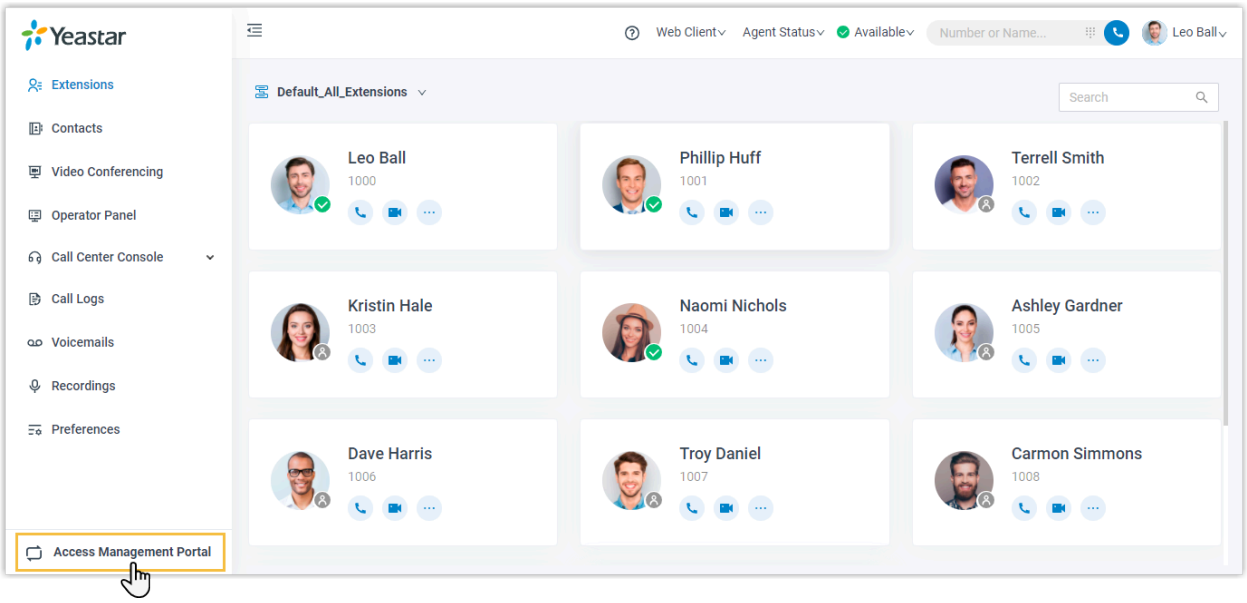
A user role is created.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. On **Extension** list, select an extension, click .
3. On the **User** page, select a role from the drop-down list of **User Role**.
4. Click **Save** and **Apply**.

Result

If specific permissions are granted to the role, after the user logs in to Linkus Web Client, the user can go to management portal, and access specific system features.



Manage User Roles

This topic describes how to edit or delete roles.

Restrictions

> **Note:**
> Only system super administrator can manage user roles.

Edit a role

After creating a role, you can edit role permissions according to your needs.

1. Log in to PBX web portal, go to **Extension and Trunk > Role**.
2. On **Role** list, select a role, click .
3. Edit the role name or change role permissions according to your needs.

   For permission details, see User Role Permissions.
4. Click **Save**.

Delete roles

If you don't need roles, you can delete them. After roles are deleted, users with the roles assigned will have no role definition.

1. Log in to PBX web portal, go to **Extension and Trunk > Role**.
2. Delete one or more roles according to your needs.
   - To delete a role, select a role, click  and **OK**.
   - To delete roles in bulk, select the checkboxes of the desired roles, click **Delete** and **OK**.

User Role Permissions

This topic describes all the available permissions that can be granted to a role.

Available permissions on Yeastar P-Series Cloud Edition are as follows:

- Extension and Trunk
- Contacts
- Call Control
- Call Features
- Reports and Recordings
- Auto Provisioning
- Messaging
- PBX Settings
- System
- Security
- Maintenance
- Integration
- Plan

Extension and Trunk

Specify the extensions that users with the role assigned can manage, and whether users can manage extension group or trunks.

Module	Permission
Extension	• **All Departments**: View and manage all the departments.
	• **All Extensions**: View and manage all the extensions.

Module	Permission
	For example, grant the permission to a human resource. If there are changes of employees, the human resource can update extensions timely. • **All the other extensions of the same Extension Groups**: Manage or send Linkus welcome emails to all the other extensions in the same extension group except the one that contains all the extensions. For example, grant the permission to a supervisor. The supervisor can view and manage his or her subordinates' extensions. • **All other extensions of the same department**: Manage or send Linkus welcome emails to all the other extensions in the same department, excluding those in the associated sub-departments. • **Specific Extensions**: Manage or send Linkus welcome emails to specific extensions. For example, grant the permission to a leader. The leader can view and manage different departments' extensions. • **Extension itself only**: Manage or send Linkus welcome emails to his or her own extension.
Extension Group	Manage extension groups.
Client Permission	Manage users' permissions to view or manage specific extensions/contacts.
Trunks	Manage trunks.

Contacts

Specify whether users with the role assigned can manage the following features:

- **Company Contacts**
- **PhoneBooks**

- **LDAP Server**

Call Control

Specify whether users with the role assigned can manage the following features:

- **Inbound Route**
- **Outbound Route**
- **AutoCLIP Route**

- **Business Hours and Holidays**
- **Emergency Number**
  - **Allow to edit Emergency Outbound Caller ID in Management Portal > Extension**: Specify whether users with the role assigned can configure emergency outbound caller ID for extensions (Path: **Management Portal > Extension and Trunk > Extension > User > Outbound Caller ID (DOD) > Emergency Outbound Caller ID**), which changes the number displayed to the called party when the extension users make an emergency call.

> **Note:**
> To specify whether the extension users can edit the emergency outbound caller ID on their own Linkus clients, go to **Management Portal > Client Permissions > Preference Settings**.

Call Features

Specify whether users with the role assigned can manage the following features:

- **Voicemail**
- **Feature Code**
- **IVR**
- **Ring Group**
- **Queue**
- **Conference**
- **Speed Dial**
- **Paging/Intercom**
- **Recording**
- **PIN List**
- **Blocked/Allowed Numbers**

Reports and Recordings

Specify users with the role assigned can view or manage which extensions' CDR and recordings, and whether users can access call reports and external chat logs.

Module	Permission
CDR	Specify users with the role assigned can view which extensions' CDR.

Module	Permission
	• **All Extensions**: View all extensions' CDR. • **All the other extensions of the same Extension Groups**: View CDR of all the other extensions of the same group except the one that contains all the extensions. • **All other extensions of the same department**: View CDR of all the other extensions in the same department, excluding those in the associated sub-departments. • **Specific Extensions**: View CDR of specific extensions.
	Specify how users with the role assigned can manage CDR. • **Download** • **Delete** • **Schedule Download**
Recording Files	Specify users with the role assigned can view which extensions' recordings. • **All Extensions**: View all extensions' recordings. • **All the other extensions of the same Extension Groups**: View recordings of all the other extensions of the same group except the one that contains all the extensions. • **All other extensions of the same department**: View recordings of all the other extensions in the same department, excluding those in the associated sub-departments. • **Specific Extensions**: View recordings of specific extensions
	Specify how users with the role assigned can manage recording files. • **Play** • **Download** • **Delete**
Call Reports	Specify whether users with the role assigned can access call reports.
External Chat Logs	Specify whether users with the role assigned can access external chat logs.

Auto Provisioning

Specify whether users with the role assigned can manage **Auto Provisioning**.

Messaging

Specify whether users with the role assigned can manage the following features:

- **Message Channel**
- **Message Queue**

PBX Settings

Specify whether users with the role assigned can manage the following features:

- **Preferences**
- **Voice Prompt**
- **SIP Settings**
- **Jitter Buffer**

System

Specify whether users with the role assigned can manage the following features:

- **Date and Time**
- **Email**
- **Storage**
- **Archive**
- **Event Notification**

Security

Specify whether users with the role assigned can manage the following features:

- **Security Rules**
- **Security Settings**

Maintenance

Specify whether users with the role assigned can manage the following features:

- **Upgrade**
- **Backup and Restore**
- **Reboot**
- **Reset**
- **Operation Logs**
- **Troubleshooting**
- **System Logs**

Integration

Specify whether users with the role assigned can manage the following features:

- **Collaboration**
- **CRM**
- **Helpdesk**
- **Speech to Text**
- **AMI**
- **API**
- **Database Grant**
- **Linkus SDK**

Plan

Specify whether users with the role assigned can buy or enable free trial of Yeastar-provided plan.

Operator Panel

Manage Operator Panel

This topic describes instructions on setting up Operator Panel for extension user.

What is Operator Panel

Operator Panel is a web-based utility integrated with the Yeastar Linkus Web Client. It is designed for employee who needs to manage and transfer a large number of calls, such as receptionist or agent manager.

For more information of managing calls on Operator Panel, see Operator Panel User Guide.

User types and permissions

There are three user types available for you to assign to an extension group member: manager, user, and custom. What they can do on Operator Panel depends on the following permission.

The following table displays the permissions available to extension group members of different user types.

> **Note:**
> By default, an extension group manager has all permissions to manage calls on Operator Panel, while the extension group users have no permission to access and use the Operator Panel.

Permission	Extension Group Manager	Extension Group User	Custom role of Extension Group
Switch group members' presence	√	√	√
Call distribution management (Redirect, Transfer, Drag and Drop operation)	√	√	√
Pick up or hang up other extensions' calls	√	√	√
Call monitoring operations (Listen, Whisper, Barge-in)	√	√	√

Permission	Extension Group Manager	Extension Group User	Custom role of Extension Group
Call parking operations (Park, Retrieve)	√	√	√
Route calls directly from IVR regardless of the IVR menu	√	√	√
Switch Business Hours and Holidays status	√	×	√
Switch extensions' recording status	√	×	√

Related information

Assign a User Type to a Group Member
View or Change Permissions for Group Members
View or Change a Member's User Type in Multiple Groups

Trunk

SIP Trunk

SIP Trunk Overview

A SIP trunk is a virtual telephone line offered by an Internet Telephony Service Provider (ITSP). Through a SIP trunk, users can make and receive calls over the internet.

Terminology

SIP

Session Initiation Protocol (SIP) is a multimedia communication protocol developed by the Internet Engineering Task Force (IETF), an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants.

ITSP

An Internet Telephony Service Provider (ITSP) is a provider of VoIP telephone service, also known as VoIP service provider.

SIP Trunk Types

Yeastar P-Series Cloud Edition supports the following SIP trunk types:

SIP Register Trunk

Registration-based SIP trunk that uses username and password for registration with SIP providers.

> **Note:**
> This type of trunk requires at least one DID number.

SIP Peer Trunk

IP-based SIP trunk that uses IP address and port of PBX for authentication.

Yeastar P-Series Cloud Edition supports the following SIP peer trunks:

- **Peer Trunk (DID Based)**: The ITSP routes calls based on DID number.

> **Note:**
> This type of trunk requires at least one DID number.

- **Peer Trunk (Port Based)**: The ITSP routes calls based on SIP registration port.

> **Note:**
> Yeastar P-Series Cloud Edition supports up to 5 port-based peer trunks.

- **Peer Trunk (Private Network)**: This trunk type can be used when the ITSP and PBX are in the same private network.

SIP Account Trunk

SIP Account Trunk is designed for connection between Yeastar P-Series Cloud Edition and other devices. Yeastar P-Series Cloud Edition will act as a VoIP account provider, the other device should register this account to connect to Yeastar P-Series Cloud Edition.

WebRTC Trunk

WebRTC (Web Real-Time Communication) trunk is used to set up WebRTC click-to-call. After you create a WebRTC trunk on the Yeastar P-Series Cloud Edition, a WebRTC inbound call link will be generated automatically, via which users can make calls to the PBX system.

SIP trunk creation methods

Create a SIP trunk by a template

Yeastar P-Series Cloud Edition supports leading ITSP across the globe, you can use the pre-configured ITSP templates included in Yeastar P-Series Cloud Edition to set up a SIP trunk quickly and easily. For more information, see Create a SIP Trunk from a Template.

> **Note:**
> Check tested and supported ITSP from ITSP partner page.

Create a general SIP trunk

If your ITSP has not undergone an interoperability test by Yeastar, or you want to set up WebRTC Click-to-Call, you can set up a general SIP trunk.

For more information, see the following topics:

- Create a SIP Register Trunk
- Create a SIP Peer Trunk
- Create a SIP Account Trunk
- Create a WebRTC trunk

SIP Trunk status

Status	Description
	Disabled.
	Unreachable.
	Registration failed. • Authentication failed. • Transport type inconsistent. • Rejected.
	Authentication failed.
	Registering.
	Registered.
	Unmonitored.
	Busy. Maximum channels reached.

Create a SIP Trunk

Create a SIP Trunk from a Template

Yeastar has tested leading ITSP across the globe and provides configuration templates for the tested and certificated ITSP. If a template is provided for your ITSP on the PBX, you can quickly create a SIP Trunk by the template.

Prerequisites

- Check if your ITSP is tested and supported by Yeastar from ITSP partner page.
- Your PBX can connect to the ITSP.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, click **Add**.
2. In the **Basic** section, configure the following settings:
   - **Name**: Enter a name to help you identify it.
   - **Trunk Status**: Select **Enabled**.
   - **Select ITSP Template**: Select your country.
   - **ITSP**: Select your ITSP.

   The trunk details are displayed automatically in the **Detailed Configuration** section.

   - If the trunk type is displayed as **Register Trunk**, configure the following settings:
     - **Username**: Enter the username provided by the ITSP.
     - **Password**: Enter the password provided by the ITSP.
     - **Authentication Name**: Optional. Authentication name is used for SIP authentication. If the ITSP provides an authentication name, enter the name.
   - If the trunk type is displayed as **Peer Trunk**, leave the settings as default.
3. **Optional:** If you have purchased DID numbers from the ITSP, click **DIDs/DDIs** tab to configure the DID numbers for the trunk.

   > **Note:**
   > If the trunk type is **Register Trunk** or **Peer Trunk (DID Based)**, this procedure is necessary.

   a. Click **Add**.
   b. In the pop-up window, configure the following settings:
      - **DID/DDI**: Enter the provided DID number.
      - **DID/DDI Name**: Optional. Enter a name to distinguish inbound calls by DID numbers.

        The name will be displayed on the called party's device when the DID number is dialed.
   c. Click **Save**.

d. To add more DID numbers, repeat **step a - c**.

For more information of DID configurations, see Configure DID Numbers on a Trunk.

4. Click **Save** and **Apply**.

Result

Go to **Extension and Trunk > Trunk** to check the trunk status on the trunk list page.

If the status shows  , the trunk is registered successfully.

For more information of SIP trunk status, see SIP Trunk status.

What to do next

- To receive inbound calls through the trunk, you need to select this trunk to one or more inbound routes. For more information, see Set up an Inbound Route.
- To make outbound calls through the trunk, you need to select this trunk to one or more outbound routes. For more information, see Set up an Outbound Route.

Create a SIP Register Trunk

This topic gives a configuration example to describe how to create a general SIP Register Trunk, which can be applied to all kinds of SIP Register Trunk.

Background information

Assume that you have bought a SIP account from the ITSP ABC, and the trunk information is displayed as below.

- **Provider domain**: abc.provider.com
- **Protocol**: SIP
- **Registration Port**: 5060
- **Transport**: UDP
- **Username**: 254258255
- **Authentication name**: 254258255
- **Password**: 05JsOmsIS54SYh

Prerequisites

- You have purchased a SIP account from an ITSP and a username and a password are offered.

- Your PBX can connect to the ITSP.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, click **Add**.
2. In the **Basic** section, configure the following settings:
   - **Name**: Enter a name to help you identify it.
   - **Trunk Status**: Select **Enabled**.
   - **Select ITSP Template**: Select **General**.
3. In the **Detailed Configuration** section, select the trunk type and enter the SIP information that is provided by the ITSP.
   - **Trunk Type**: Select **Register Trunk**.
   - **Transport**: Select the transport provided by the ITSP. In this example, select **UDP**.
   - **Hostname/IP**: Enter the domain name or IP address of the ITSP. In this example, enter *abc.provider.com*.
   - **Port**: Enter the provided registration port. In this example, enter *5060*.
   - **Domain**: Enter the domain in SIP URI of a specific header like From, To header. In this example, enter *abc.provider.com*.

     > **Note:**
     > If the domain is not provided by ITSP, enter the same value as **Hostname/IP**.

   - **Username**: Enter the provided user name. In this example, enter *254258255*.
   - **Password**: Enter the provided password. In this example, enter *05JsOms-IS54SYh*.
   - **Authentication Name**: Enter the provided authentication name. In this example, enter *254258255*.

     > **Note:**
     > In most cases, authentication name is the same as the user name.

   - **Enable Outbound Proxy**: Optional. If the trunk is configured to use an outbound proxy server, when users make outbound calls through this trunk, all the SIP packets will be sent to the outbound proxy server.

     > **Note:**

> Contact your ITSP to check if outbound proxy is supported, then configure outbound proxy settings under the ITSP's guidance.

4. Click **DIDs/DDIs** tab to configure the DID numbers for the trunk.
   a. Click **Add**.
   b. In the pop-up window, configure the following settings:
      - **DID/DDI**: Enter the DID number provided by the ITSP.
      - **DID/DDI Name**: Optional. Enter a name to distinguish inbound calls by DID numbers.

        The name will be displayed on the called party's device when the DID number is dialed.
   c. Click **Save**.
   d. To add more DID numbers, repeat **step a - c**.

   For more information of DID configurations, see Configure DID Numbers on a Trunk.
5. **Optional:** Click **Advanced**, **Inbound Caller ID Reformatting**, **Outbound Caller ID**, or **SIP Headers** tab to configure other settings.
6. Click **Save** and **Apply**.

Result

Go to **Extension and Trunk > Trunk** to check the trunk status on the trunk list page.

If the status shows  , the trunk is registered successfully.

For more information of SIP trunk status, see SIP Trunk status.

What to do next

- To receive inbound calls through the trunk, you need to select this trunk to one or more inbound routes. For more information, see Set up an Inbound Route.
- To make outbound calls through the trunk, you need to select this trunk to one or more outbound routes. For more information, see Set up an Outbound Route.

Create a SIP Peer Trunk

This topic gives a configuration example to describe how to create a general SIP Peer Trunk, which can be applied to all kinds of SIP Peer Trunk.

Background information

Assume that you have bought a SIP account from the ITSP ABC, and the trunk information is displayed as below.

- **Provider domain**: abc.provider.com
- **Protocol**: SIP
- **Registration Port**: 5060
- **Transport**: UDP

Prerequisites

- You have purchased a SIP account from an ITSP and no username and password is offered but only a domain name or IP address.
- Your PBX can connect to the ITSP.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, click **Add**.
2. In the **Basic** section, configure the following settings:
   - **Name**: Enter a name to help you identify it.
   - **Trunk Status**: Select **Enabled**.
   - **Select ITSP Template**: Select **General**.
3. In the **Detailed Configuration** section, select the trunk type and enter the SIP information that is provided by the ITSP.
   - **Trunk Type**: Select the desired type of **Peer Trunk**.
     - **Peer Trunk (DID Based)**: If the ITSP routes calls based on the DID number, choose this option.

       > **Note:**
       > If this trunk type is selected, it requires at least one DID rule.

     - **Peer Trunk (Port Based)**: If the ITSP routes calls based on the SIP registration port, choose this option.
     - **Peer Trunk (Private Network)**: If the ITSP and the PBX are in the same private network, choose this option.

   The **Static IP Address** and **Port** of the PBX will be displayed on the web page.

- **Transport**: Select the transport provided by the ITSP. In this scenario, select **UDP**.
- **Hostname/IP**: Enter the domain name or IP address of the ITSP. In this scenario, enter *abc.provider.com*.
- **Port**: Enter the provided registration port. In this scenario, enter *5060*.
- **Domain**: Enter the domain in SIP URI of a specific header like From, To header. In this example, enter *abc.provider.com*.

> **Note:**
> If the domain is not provided by ITSP, enter the same value as **Hostname/IP**.

4. **Optional:** If you have purchased DID numbers from the ITSP, click **DIDs/DDIs** tab to configure the DID numbers for the trunk.

> **Note:**
> If the selected trunk type is **Peer Trunk (DID Based)**, this procedure is necessary.

   a. Click **Add**.
   b. In the pop-up window, configure the following settings:
      - **DID/DDI**: Enter the provided DID number.
      - **DID/DDI Name**: Optional. Enter a name to distinguish inbound calls by DID numbers.

        The name will be displayed on the called party's device when the DID number is dialed.
   c. Click **Save**.
   d. To add more DID numbers, repeat **step a - c**.

   For more information of DID configurations, see Configure DID Numbers on a Trunk.
5. Click **Save** and **Apply**.

Result

Go to **Extension and Trunk > Trunk** to check the trunk status on the trunk list page.

If the status shows , the trunk is registered successfully.

For more information of SIP trunk status, see SIP Trunk status.

What to do next

- To receive inbound calls through the trunk, you need to select this trunk to one or more inbound routes. For more information, see Set up an Inbound Route.
- To make outbound calls through the trunk, you need to select this trunk to one or more outbound routes. For more information, see Set up an Outbound Route.

Create a SIP Account Trunk

A SIP Account is used for the other device to register with Yeastar P-Series Cloud Edition. In this way, Yeastar P-Series Cloud Edition and the other device are connected. This topic describes how to create a SIP Account Trunk on Yeastar P-Series Cloud Edition.

Prerequisites

To connect a third-party device with Yeastar P-Series Cloud Edition by a SIP Account Trunk, you need to make sure that there is no duplicate extension numbers on both sides.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, click **Add**.
2. In the **Basic** section, configure the following settings:
   - **Name**: Enter a name to help you identify it.
   - **Trunk Status**: Select **Enabled**.
   - **Select ITSP Template**: Select **General**.
3. In the **Detailed Configuration** section, configure the following settings:

> **Note:**
> You can leave the default SIP information or edit the information according to your needs.

Setting	Description
Trunk Type	Select **Account Trunk**.
Transport	Select a transport. The following options are supported: • **UDP** • **TCP** **Note:**

Setting	Description
	By default, SIP TCP port is disabled. If you select this option, go to **PBX Settings > SIP Settings > General > Basic**, enable **SIP TCP Port**, then reboot the PBX to make it take effect. • **TLS**
Username	Enter a username for the SIP account.
Password	Enter a password for the SIP account.
Authentication Name	Retain the default authentication name or enter a custom one. **Note:** If the device to be connected does NOT support the **Authentication Name** field, select **Use User Name as Account Trunk's Authentication Name**.

4. **Optional:** Click **Advanced**, **Inbound Caller ID Reformatting**, **Outbound Caller ID**, or **SIP Headers** to configure other settings.
5. Click **Save** and **Apply**.

What to do next

- Register the SIP Account Trunk on the third-party software or device. You need to provide the following information:
  - SIP Account Trunk details
  - Domain name of PBX
  - SIP registration port: The default port is 5060 on Yeastar P-Series Cloud Edition
- To receive inbound calls through the trunk, you need to select this trunk to one or more inbound routes. For more information, see Set up an Inbound Route.
- To make outbound calls through the trunk, you need to select this trunk to one or more outbound routes. For more information, see Set up an Outbound Route.

Result

Go to **Extension and Trunk > Trunk** to check the trunk status on the trunk list page.

If the SIP Account Trunk is successfully registered on the third-party software or device, the trunk status will show , which also indicates that the two devices are connected.

For more information of SIP trunk status, see SIP Trunk status.

Manage SIP Trunks

After you create SIP trunks, you can edit or delete the SIP trunks.

Edit a SIP trunk

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**.
2. On the Trunk list page, select a trunk and click .
3. Click the desired tab to edit the relevant settings.
4. Click **Save** and **Apply**.

Delete SIP trunks

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**.
2. To delete a SIP trunk, do the followings:
    a. Click beside the trunk.
    b. Click **Yes** in the pop-up dialog box to confirm.
3. To delete multiple SIP trunks, do the followings:
    a. Select checkboxes of the desired trunks.
    b. Click **Delete**.
    c. Click **Yes** in the pop-up dialog box to confirm.

Export and Import SIP Trunks

The SIP trunks configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired SIP trunks in the exported file, and import the file to PBX again. This topic describes how to export and import SIP trunks.

Background information

Only Peer Trunks and Register Trunks can be imported.

Export all SIP trunks

You can export all SIP trunks to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**.
2. Click **Export**.

A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Trunk Parameters.

Import SIP trunks

We recommend that you export SIP trunks data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Trunk Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The trunks in the CSV file will be displayed in the **Trunk** list.

Related information

Import and Export -FAQ

SIP Trunk Settings

This topic describes all the settings on a SIP trunk for reference.

Basic settings

Basic

Setting	Description
Name	Give this trunk a name to help you identify it.
Trunk Status	Enable or disable the trunk.

Setting	Description
Select ITSP Template	Select the country of your ITSP. **Note:** If no SIP trunk template is provided for your ITSP, select **General**.
ITSP	Select your ITSP from the list of certified SIP trunk providers.

Detailed Configuration

Setting	Description
Trunk Type	Select a trunk type: • **Register Trunk** • **Peer Trunk (DID Based)** • **Peer Trunk (Port Based)** **Note:** Yeastar P-Series Cloud Edition supports up to 5 **Peer Trunk (Port Based)**. • **Peer Trunk (Private Network)** • **Account Trunk**
Register Trunk	
Transport	Select the transport that is provided by the ITSP. **Note:** If you select **TCP**, make sure SIP TCP Port is enabled (Path: **PBX Settings > SIP Settings > General > Basic > SIP TCP Port**.
Hostname/IP	Enter the IP address or the domain of the ITSP.
Port	Enter the SIP port provided by the ITSP.
Domain	Enter the domain in SIP URI of a specific header like From, To header. **Note:** If the domain is not provided by ITSP, enter the same value as **Hostname/IP**.

Setting	Description
Username	Enter the username to register to the ITSP.
Password	Enter the password that is associated with the username.
Authentication Name	Enter the authentication name to register to the ITSP.
Enable Outbound Proxy	If the trunk is configured to use an outbound proxy server, when users make outbound calls through this trunk, all the SIP packets will be sent to the outbound proxy server. **Note:** Contact your ITSP to check if they support outbound proxy, then configure outbound proxy settings under their guidance.
Peer Trunk (DID Based/Port Based/Private Network)	
Transport	Select the transport that is provided by the ITSP. **Note:** If you select **TCP**, make sure SIP TCP Port is enabled (Path: **PBX Settings > SIP Settings > General > Basic > SIP TCP Port**.
Static IP Address	The Static IP Address is assigned automatically, this field can not be edited.
Port	The port is assigned automatically, this field can not be edited..
Hostname/IP	Enter the IP address or the domain of the ITSP.
Port	Enter the SIP port provided by the ITSP.
Domain	Enter the domain in SIP URI of a specific header like From, To header. **Note:** If the domain is not provided by ITSP, enter the same value as **Hostname/IP**.
Account Trunk	
Transport	Select the transport for a third-party device to register with.

Setting	Description
	Note: If you select **TCP**, make sure SIP TCP Port is enabled (Path: **PBX Settings > SIP Settings > General > Basic > SIP TCP Port**.
Username	Specify a username for the trunk. **Note:** The username is regarded as the trunk number.
Password	Specify a password that is associated with the username.
Authentication Name	Specify an authentication name for a third-party device to register with.

Advanced settings

The advanced settings of VoIP trunk require professional knowledge of SIP protocol. Incorrect configurations may cause calling issues. It is wise to leave the default settings provided on the SIP trunk page. However, for a few fields, you need to change them to suit your situation.

The following settings are included on the **Advanced** page.

- Codec Setting
- VoIP Setting
- Call Restriction

Codec Setting

Each newly created SIP trunk has a default preferred codec list. However, the default codec list may not match the codecs supported by your ITSP. To maximize the quality of calls and the amount of bandwidth used for calls, you can configure your preferred codec list to match the settings that your ITSP supports.

Yeastar P-Series Cloud Edition supports the following codecs:

- u-law
- a-law
- G729A

- GSM
- H264
- H261
- H263
- H263P
- iLBC
- G722
- G726
- SPEEX
- ADPCM
- MPEG4
- VP8
- Opus

VoIP Setting

Setting	Description
DTMF Mode	Set the default mode for sending DTMF tones. • **RFC4733 (RFC2833)**: DTMF will be carried in the RTP stream in different RTP packets rather than the audio signal. • **Info**: DTMF will be carried in the SIP info messages. • **Inband**: DTMF will be carried in the audio signal. • **Auto**: The PBX will detect if the device supports **RFC4733(RFC2833)** DTMF. If **RFC4733(RFC2833)** is supported, PBX will choose **RFC4733(RFC2833)**, or the PBX will choose **Inband**.
Qualify	Enable this option to send SIP OPTION packet to SIP device to check if the device is up.
Enable SRTP	Enable or disable SRTP (encrypted RTP) for the trunk.
T.38 Support	Enable or disable T.38 fax for this trunk. Enabling T.38 will add the performance cost. We suggest that you disable T.38.
Inband Progress	This Inband Progress setting applies to the extensions which make calls through this trunk. **Note:**

Setting	Description
	To configure global Inband Progress setting, you need to contact Yeastar support to configure a custom config file. • **Check this option**: PBX will send a 183 Session Progress to the extension when told to indicate ringing and will immediately start sending ringing as audio. • **Uncheck this option**: PBX will send a 180 Ringing to the extension when told to indicate ringing and will NOT send it as audio.
Ignore 183 Message without SDP	This option determines the way PBX handles 183 messages without SDP. • **Check this option**: PBX will not forward 183 messages that don't contain SDP. • **Uncheck this option**: PBX will process all the 183 messages without SDP to those with SDP and forward them.
Dedicated Trunk	If your SIP trunk provider requires a dedicated internal IP address for connection, you need to enable this option and contact your service provider to configure the related network settings. **Note:** The option is only available for the following trunk types: • Register trunk • Peer Trunk (DID Based) • Peer Trunk (Port Based)
Authentication Error SIP Code	Specify the SIP error code returned by the ITSP to indicate an authentication error. **Note:** • This option is only available for register trunk. • You can enter up to 10 error codes; Use ; to separate multiple error codes.

Setting	Description
Authentication Registration Attempts	Set the number of registration attempts for the PBX after receiving the authentication error SIP code from the ITSP. **Note:** • This option is only available for register trunk. • The registration attempts defined here include the initial attempt without authentication information.
Global Registration Retry Interval (s)	Set the global interval time (in seconds) for retrying registration after receiving any SIP error code (non-200 status code) from ITSP. **Note:** This option is only available for register trunk.

Call Restriction

Setting	Description
Call Restriction Type	Specify based on which type of calls to restrict the max concurrent call number of this trunk. • **Outbound Call**: Only outbound calls will be restricted. • **All**: Both outbound calls and inbound calls will be restricted.
Maximum Concurrent Calls	Specify the maximum number of concurrent calls allowed in this trunk. The default is Unlimited.

DIDs/DDIs

Direct Inward Dialling (DID), also called Direct Dial-in (DDI), is a service offered by telephone companies. For more information of DID concepts, see DID Number Overview.

- DID numbers are usually configured on inbound routes to distinguish inbound calls.

  For more information, see Route Inbound Calls based on DID Numbers.
- For more instructions on configuring the DID numbers, see Configure DID Numbers on a Trunk.

Inbound Caller ID Reformatting

When a user calls in the PBX, the trunk provider may send a caller ID that is inconvenient for you to redial directly.

In this case, you can reformat inbound caller ID based on a trunk. The caller ID will be reformatted before it is sent to the called party.

For more information, see Reformat Inbound Caller ID based on a Trunk.

Outbound Caller ID

Outbound caller ID is the phone number or name that is displayed on the called party's device.

You can set up a global outbound caller ID for a trunk or assign caller IDs for extension users.

> **Note:**
> By default, each trunk has a default phone number that will be displayed on the called party's device. Outbound Caller ID configuration requires support from the trunk provider. Contact your trunk provider first before you configure Outbound Caller ID, or the settings won't take effect and outbound calls may fail.

If you set the caller ID number, when users make outbound calls through this trunk, the called party will see this caller ID number instead of the calling party's number.

For more information of outbound caller ID configurations, see Customize Outbound Caller IDs

SIP Headers

The SIP Headers settings require professional knowledge of SIP protocol. Incorrect configurations may cause calling issues. It is wise to leave the default settings provided on the SIP trunk page. However, for a few fields, you need to change them to suit your situation.

The following settings are included on the **SIP Headers** page.

- Inbound Parameters
- Outbound Parameters
- Other Settings

Inbound Parameters

Parameter	Description
Get Caller ID From	Decide from which header field will the trunk retrieve Caller ID. • **Follow System** The trunk will follow the global Get Caller ID From setting. • **From** • **Contact** • **Remote-Party-ID** • **P-Asserted Identify** • **P-Preferred-Identity**
Get DID From	Different devices or providers may contain DID numbers in different SIP headers. When an inbound call through a SIP trunk reaches the PBX, the PBX needs to retrieve a correct DID number, or the call will fail. Adjust the setting after analysis of the SIP packets sent from the trunk provider. The following SIP headers are available to select: • **Follow System** The trunk will follow the global Get DID From setting. • **To** • **Invite** • **Diversion** • **Remote-Party-ID** **Note:** If this option is selected, but the SIP provider doesn't support Remote Party ID, the PBX will retrieve DID from INVITE header. • **P-Asserted Identify** • **P-Called-Party-ID** • **P-Preferred-Identity**

Outbound Parameters

For outbound calls, you can define the parameters included in the following SIP INVITE headers:

Parameter	Description
From User Part	Define the caller ID that will be used of a SIP From header. For more information, see Options of outbound parameters.

<table>
<tr><th>Parameter</th><th>Description</th></tr>
<tr><td>From Display Name Part</td><td>Define the caller ID name that will be used of a SIP From header.
For more information, see Options of outbound parameters.</td></tr>
<tr><td>From Host Part</td><td>Important:
This parameter is only available for Peer Trunks.
Set the parameter according to the requirements of your SIP trunk provider, otherwise, it may cause call issues.
Define the domain or IP address to be used in the From field of a SIP INVITE header.
Default: Use the domain or IP address configured in the Domain field when creating the Peer Trunk.
Custom: Use a custom domain or IP address. You can enter the custom value in the field next to the Custom drop-down list.</td></tr>
<tr><td>To Host Part</td><td>Important:
This parameter is only available for Peer Trunks.
Set the parameter according to the requirements of your SIP trunk provider, otherwise, it may cause call issues.
Define the domain or IP address to be used in the To field of a SIP INVITE header.
Default: Use the domain or IP address configured in the Domain field when creating the Peer Trunk.
Custom: Use a custom domain or IP address. You can enter the custom value in the field next to the Custom drop-down list.</td></tr>
<tr><td>Diversion</td><td rowspan="3">Optional: Define other parameters of a SIP INVITE header as needed.
For more information, see Options of outbound parameters.</td></tr>
<tr><td>Remote-Party-ID</td></tr>
<tr><td>P-Asserted-Identity</td></tr>
</table>

Parameter	Description
P-Preferred-Identity	

Note:
For different types of SIP trunk, the optional items are different.

Table 14. Options of outbound parameters

Option	Description
[Default]	• For the **Diversion** outbound parameter, the system uses the original call destination number (DID number) as the parameter value when the call diversion occurs. • For the **Remote-Party-ID**, **P-Asserted-Identity** and **P-Preferred-Identity** outbound parameter, the system selects a parameter by the following priority from top to bottom: ◦ Outbound Route Outbound Caller ID ◦ Extension's Outbound Caller ID in Trunk ◦ Trunk Outbound Caller ID ◦ Trunk Username ◦ Extension Caller ID ◦ The Originator Caller ID
[None]	Do not send the parameter with the SIP INVITE packet.
Extension Caller ID	The caller ID configured on the extension.
Trunk Outbound Caller ID	The global outbound caller ID for the trunk (**Trunk > Outbound Caller ID > General**).
Extension's Outbound Caller ID in Trunk	The extension's associated outbound caller ID with the trunk.
Outbound Route Outbound Caller ID	The outbound caller ID configured on the outbound route that is used for the outbound calls.
Originator Caller ID	The Caller ID of the call originator (the first caller in the case that the call is transferred). • If the call originator is an external number, the external number will be taken. • If the call originator is an extension, the priority order will be Extension Outbound Caller ID → [Default].
Trunk Username	The username configured on the trunk.
Custom	Define a custom value.

Other Settings

Setting	Description
User Agent	If the ITSP requires User Agent for authentication, enter the User Agent information that is provided by the ITSP.
Realm	Realm is a string displayed to users so they know which username and password to use. **Note:** If you don't know what to fill in, contact your service provider for further instructions.
Send Privacy ID	Whether to send the Privacy ID in SIP header or not. The default is unchecked.
User Phone	Whether to add the parameter `user=phone` as a request line in the header field of the SIP INVITE packet. **Note:** Enable this option only when the SIP provider requires.
100rel	Configure 100rel for this trunk. • **Required**: 100rel is required for this trunk. • **Supported**: 100rel is supported by this trunk. • **Disabled**: 100rel is disabled for this trunk.
Maxptime	Select the value of the maxptime used when the PBX sends the INVITE packet. **Note:** If you select **[Default]**, PBX will send a corresponding maxptime value according to the codec that is used for the outbound call.
Support P-Early-Media	Set whether the P-Early-Media field is included in the INVITE packet.

WebRTC Trunk

WebRTC Trunk Overview

Yeastar P-Series Cloud Edition supports WebRTC trunk. Unlike traditional SIP trunks, WebRTC trunk uses WebRTC (Web Real-Time Communication) technology to implement real-time audio and video communication on web page. With a simple click on a call link provided by WebRTC trunk, your customers can make free calls directly from their browser to your company's phone system without the need for additional plug-ins or software. In addition, the audio call can be seamlessly switched to a video call for face-to-face communication, improving the customer service experience.

Requirements and restrictions

Requirements

- **Plan**: If you need video calls, **Ultimate Plan** is required.
- **Firmware**: The version of Yeastar P-Series Cloud Edition is 84.11.0.22 or later.

Restrictions

- You can create up to **5** WebRTC trunks.
- A WebRTC trunk supports up to **10** concurrent calls.

Supported web browsers

The following table shows the compatible web browsers.

Web browser	Version
Google Chrome	Chrome 87 or later
Microsoft Edge	Edge 87 or later
Opera	Opera 72 or later

Related information

Set up WebRTC Click-to-Call

Set up WebRTC Click-to-Call

To implement WebRTC click-to-call, you need to create a WebRTC trunk on PBX to obtain a WebRTC call link, then embed the call link into your web page. In this way, your website visitors can make calls to your company's phone system with a click.

Prerequisites

If you need video calls, make sure that you have subscribed to **Ultimate Plan**.

Procedure

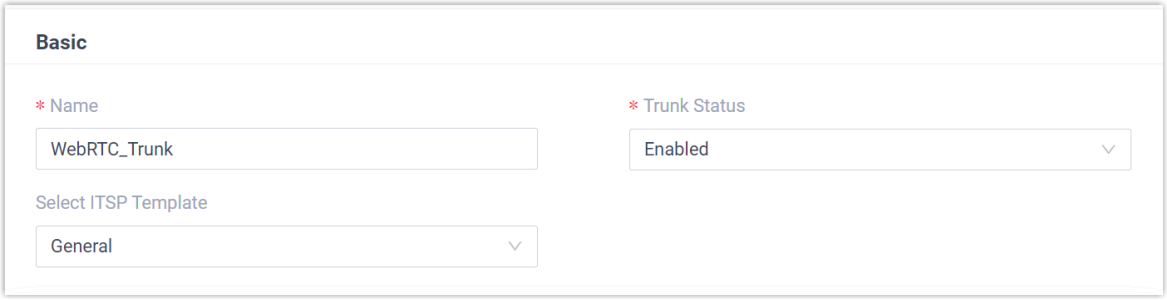
1. Create a WebRTC trunk
2. Set WebRTC inbound call destination
3. Test WebRTC Click-to-Call

Video Tutorial

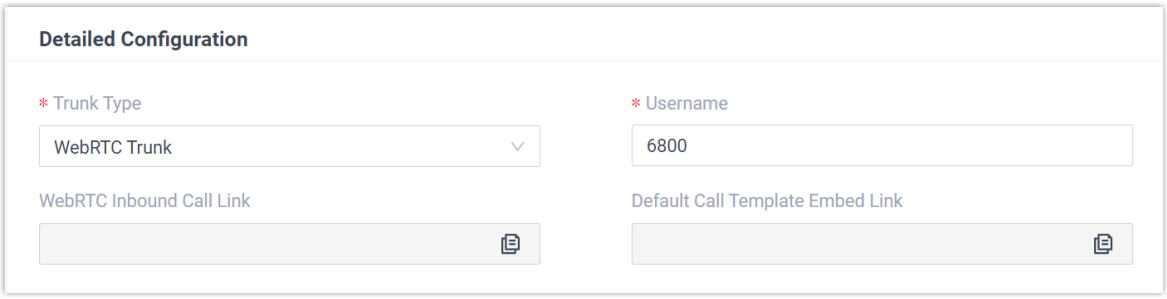
Create a WebRTC trunk

Create a WebRTC trunk to obtain a WebRTC call link.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, then click **Add**.
2. In the **Basic** section, complete the following settings.



   - **Name**: Enter a name to help you identify it.
   - **Trunk Status**: Select **Enabled**.
   - **Select ITSP Template**: Select **General**.
3. In the **Detailed Configuration** section, complete the following settings.



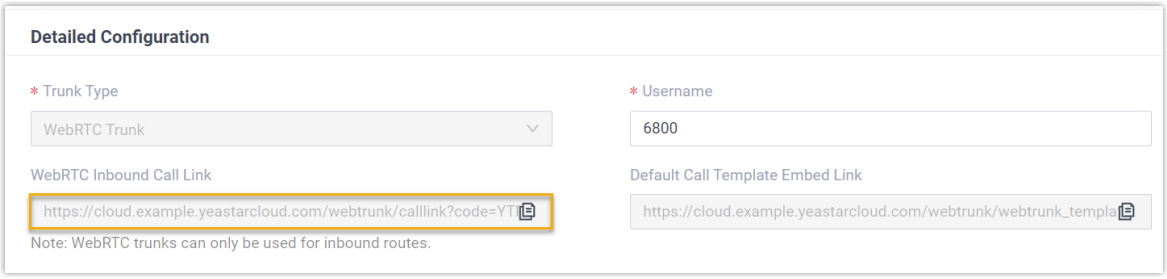
- **Trunk Type**: Select **WebRTC Trunk**.
- **Username**: Enter a username for the trunk.

4. **Optional:** Click the **Advanced** tab to configure advanced WebRTC trunk settings.
   a. In the **Codec Settings** section, configure your preferred codecs.
   b. In the **Call Restriction** section, set the maximum number of concurrent calls for the trunk.

> **Note:**
> The default number is **5**, you can set up to **10** concurrent calls.

5. Click **Save** and **Apply**.

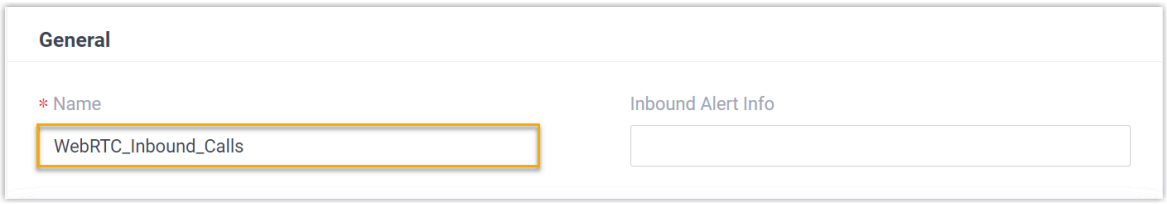
A WebRTC trunk is displayed in the trunk list with the trunk status showing ; A WebRTC call link is generated and displayed in the **WebRTC Inbound Call Link** field of the trunk.



Set WebRTC inbound call destination

Set up an inbound route for the WebRTC trunk to route the WebRTC inbound calls to a desired destination.

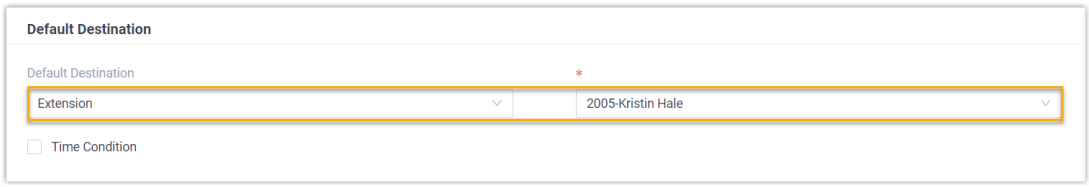
1. Go to **Call Control > Inbound Route**, then click **Add** to add an inbound route.
2. In the **General** section, set a name for the inbound route.



3. In the **Trunk** section, select the WebRTC trunk from **Available** box to **Selected** box.

> **Note:**
> You can NOT select both WebRTC trunk and other types of trunk for an inbound route at the same time.

4. In the **Default Destination** section, set destination(s) for the WebRTC inbound calls.
   - If you want the inbound calls to always be routed to a destination, set the desired destination in the **Default Destination** drop-down list.



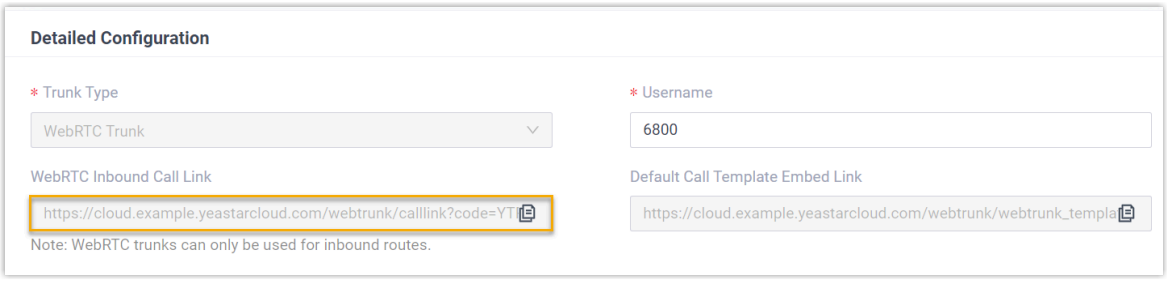
   - If you want the inbound calls to be routed to different destinations based on time, do as follows:
     a. Select the checkbox of **Time Condition**.
     b. In the **Time-based Routing Mode** drop-down list, select a time-based mode.
     c. Configure the corresponding destinations based on time.

     For more information of time-based inbound call routing, see the following topics:
     - Route Inbound Calls based on Global Business Hours
     - Route Inbound Calls based on Department Hours
     - Route Inbound Calls based on Employee Hours
5. Click **Save** and **Apply**.

Test WebRTC Click-to-Call

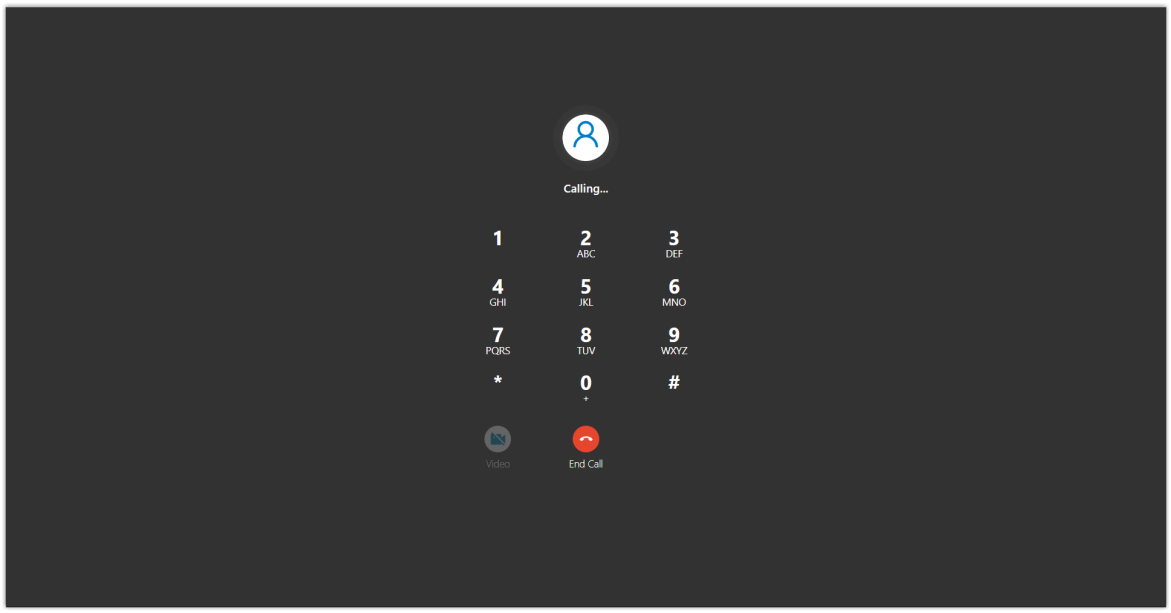
Test if the WebRTC inbound call can be routed to the given destination.

1. Go to **Extension and Trunk > Trunk**, click beside the WebRTC trunk.
2. In the **Detailed Configuration** section, copy the **WebRTC Inbound Call Link**.



3. Open a browser tab, paste the inbound call link in the address bar, then press **Enter**. A dialpad is displayed on the web page, and a call is placed to the pre-set call destination.

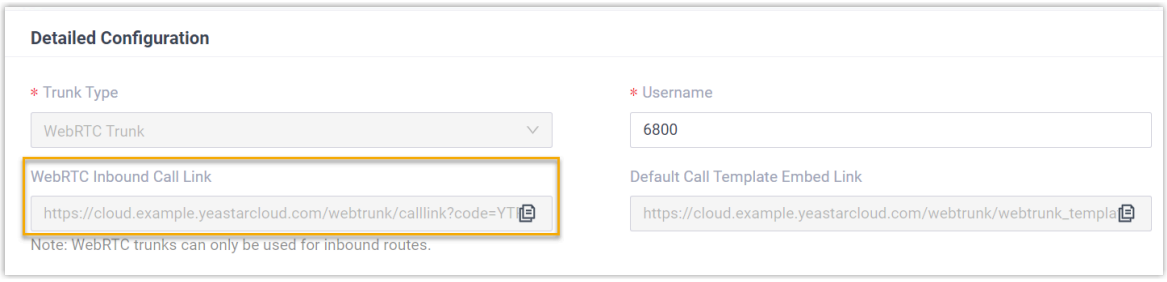
> **Tip:**
>
> During the call, you can switch to a video call by clicking at the bottom-left corner on the web page.



What to do next

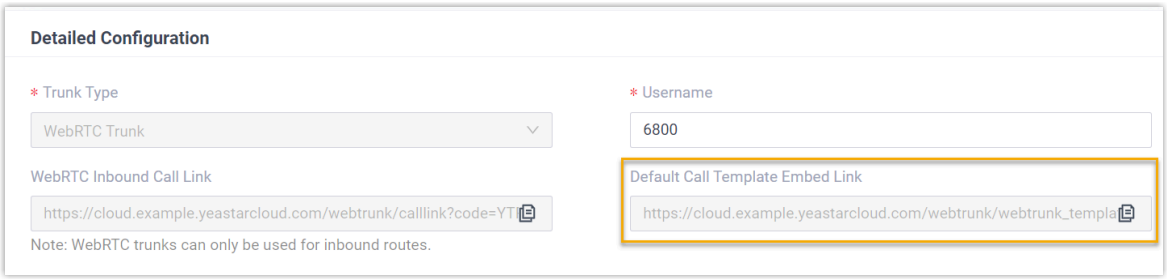
Embed the WebRTC inbound call link into your web page in either of the following ways:

- Customize a call button on your website, and set the button link to the WebRTC call link.

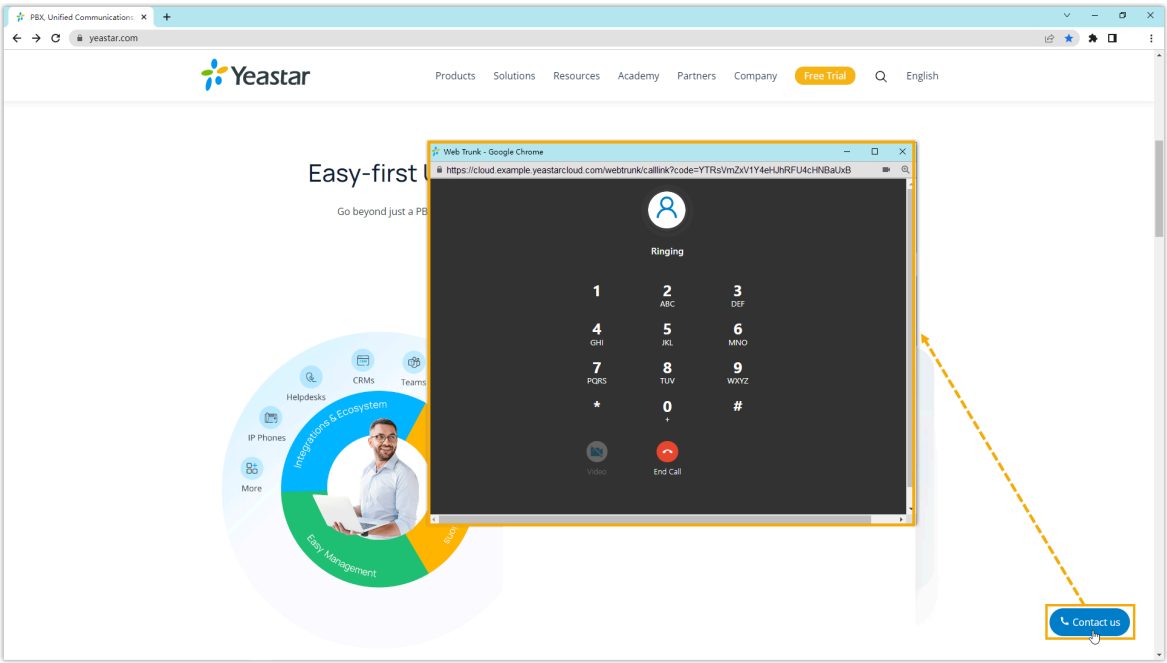


- Utilize the default call button template provided in the WebRTC trunk by referencing the embed link in your website using the following script:

```
<script src="{Default Call Template Embed Link}"></script>
```



The figure below shows the generated call button and the click-to-call effect:



Call Control

Emergency Calling

Emergency Calling Overview

This topic describes concepts that you need to know before managing emergency calling, including requirements and restrictions, basic emergency calling, and enhanced emergency calling.

Requirements

To make an emergency call, you should make sure the following requirements are met:

- IP phones or soft phones must be registered to Yeastar P-Series Cloud Edition.
- At least one trunk should be configured for an emergency number.

Basic emergency calling

The basic emergency service only connects a caller to the local Public Safety Answering Point (PSAP), but no location is provided. Emergency callers must be ready to provide their location information for the PSAP. PSAP then arranges appropriate emergency response after communicating with the callers.

For more information, see Set up Basic Emergency Calling.

Enhanced emergency calling

Enhanced emergency service is only available for specific countries and regions, such as E911 in North America, E112 in continental Europe, E999 in England, etc.

For an enhanced emergency call, PSAP can immediately pinpoint the caller's location based on the calling number.

> **Important:**
> For wireless IP phones and softphones (such as Linkus), the emergency caller's location can only be determined by the Emergency Outbound Caller ID configured on the PBX.

For more information, see Set up Enhanced Emergency Calling.

Terminology

The following list defines the key terminology for enhanced emergency calling.

PSAP

A Public Safety Answering Point (PSAP) is responsible for receiving emergency calls and arranging appropriate emergency response, such as dispatching a police, fire, or ambulance team.

ERL

An Emergency Response Location (ERL) is a specific geographic location to which an emergency response team may be dispatched. To provide the PSAP with the emergency caller's precise location, you may need to set multiple ERLs.

ELIN

An Emergency Location Identification Number (ELIN) is the phone number (Caller ID), which is associated with an ERL. When an emergency call is made, the ELIN is displayed on the PSAP side so that they can match the caller ID with the ERL.

> **Note:**
> ELIN is also helpful for PSAP to call the emergency caller back in case the call is disconnected.

Examples of ERL/ELIN mapping:

- **One ERL for each building**

  All the users in the same building are associated with the same ELIN.

ELIN	ERL
6085225672	No. 63-2 Wanghai Road, 2nd Software Park, Xiamen
6085225673	No. 63-3 Wanghai Road, 2nd Software Park, Xiamen

- **One ERL for each building floor**

  All the users on the same floor of a building are associated with the same ELIN.

ELIN	ERL
6085225682	5/F, No. 63-2 Wanghai Road, 2nd Software Park, Xiamen
6085225683	4/F, No. 63-2 Wanghai Road, 2nd Software Park, Xiamen

- **One ERL for each room**

  Each user of a room has a unique ELIN.

ELIN	ERL
6085225692	Room3005, No.1 Guanri Road, Software Park Siming District Xiamen
6085225693	Room3006, No.1 Guanri Road, Software Park Siming District Xiamen

Set up Basic Emergency Calling

To ensure that users can make emergency calls for help when an accident occurs, you need to set up emergency calling in Yeastar P-Series Cloud Edition. This topic describes how to set up basic emergency calling in Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number**, click **Add**.
2. In the **Name** field, specify a name to help you identify it.
3. In the **Emergency Number** field, enter the emergency number.
4. Leave the **Emergency Outbound Caller ID Priority** field as the default setting.

> **Note:**
> - **Emergency Outbound Caller ID Priority** setting is typically for enhanced emergency calling, this setting will not affect basic emergency calling.
> - For basic emergency calling, you don't need to set Emergency Outbound Caller ID for extensions and trunks.

5. Under the **Outbound Rules** tab, configure trunks for emergency calls.

> **Note:**

> Emergency calls have the highest priority. If the selected trunk is occupied, PBX will terminate the ongoing call, and place the emergency call.

a. Click **Add**.
b. In the drop-down list of **Trunk**, select a trunk.
c. Leave the **Trunk's Emergency Outbound Caller ID** field blank.

> **Note:**
> Do not set emergency outbound caller ID for basic emergency calling, or the emergency calls may fail.

d. If the ITSP requires a prefix to place outbound calls, enter the provided prefix number in the **Prepend for Emergency Number** field.

   The prefix will be automatically added at the beginning of the dialed emergency number. For example, if you set the field to `1234`, when a user dials 911, the PBX system will add 1234 to the emergency number and call out 1234911.

> **Important:**
> Carefully configure the prefix according to the ITSP's requirements, or the emergency calls may fail.

e. **Optional:** Repeat **step a** - **step d** to add another trunk.

> **Note:**
> If the first trunk cannot work properly, the PBX will use the second trunk to make calls.

6. Click **Save** and **Apply**.

What to do next

After setting up an emergency calling, you may need to consider the following configurations:

- Set up a Route for PSAP Callbacks
- Add an Emergency Notification Contact

Set up Enhanced Emergency Calling

To ensure that users can make emergency calls for help when an accident occurs, you need to set up emergency calling in Yeastar P-Series Cloud Edition. This topic describes how to set up enhanced emergency calling in Yeastar P-Series Cloud Edition.

Prerequisites

Purchase enhanced emergency service from an Internet Telephony Service Provider (ITSP).

ITSP will provide DID numbers that are associated with your locations. DID number is also called Emergency Location Identification Number (ELIN).

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number**, click **Add**.
2. In the **Name** field, specify a name to help you identify it.
3. In the **Emergency Number** field, enter the emergency number.
4. In the **Emergency Outbound Caller ID Priority** field, select which outbound caller ID will be sent to the Public Safety Answering Point (PSAP) in priority when an emergency call is made.
   - **Trunk's Emergency Outbound Caller ID**: Select this option if you want to set a common ELIN for all extension users. PSAP receives the trunk's emergency outbound caller ID no matter who makes the emergency call, which indicates PSAP receives a common location information.
   - **Extension's Emergency Outbound Caller ID**: Select this option if you want to assign ELINs for individual users.
     - Extension users with specific ELINs are associated with their respective locations.
     - Extension users without specific ELINs share a common ELIN (the trunk's emergency outbound caller ID) and are associated with a common location.
5. Under the **Outbound Rules** tab, configure trunks for emergency calls.

   > **Note:**
   > Emergency calls have the highest priority. If the selected trunk is occupied, PBX will terminate the ongoing call, and place the emergency call.

   a. Click **Add**.
   b. In the drop-down list of **Trunk**, select a trunk.

c. In the **Trunk's Emergency Outbound Caller ID** field, enter the Emergency Location Identification Number (ELIN) that you have purchased from the trunk provider.

d. If the ITSP requires a prefix to place outbound calls, enter the provided prefix number in the **Prepend for Emergency Number** field.

The prefix will be automatically added at the beginning of the dialed emergency number. For example, if you set the field to `1234`, when a user dials 911, the PBX system will add 1234 to the emergency number and call out 1234911.

> **Important:**
> Carefully configure the prefix according to the ITSP's requirements, or the emergency calls may fail.

e. **Optional:** Repeat **step a** - **step d** to add another trunk.

> **Note:**
> If the first trunk cannot work properly, the PBX will use the second trunk to make calls.

6. Click **Save** and **Apply**.

Assign ELINs for individual users

To provide the PSAP with the emergency caller's precise location, you may need to purchase multiple ELINs and assign these ELINs to extension users.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click to edit the desired extension.
2. On the extension **User** page, scroll down the page, enter the ELIN in the **Emergency Outbound Caller ID** field.
3. Click **Save** and **Apply**.

After the user dials an emergency number, the PSAP will locate the specific geographic location of the user by the extension user's ELIN.

What to do next

After setting up an emergency calling, you may need to consider the following configurations:

- Set up a Route for PSAP Callbacks
- Add an Emergency Notification Contact

Set up a Route for PSAP Callbacks

In case that the emergency caller is not available to answer the returned call from PSAP, you can set up an inbound route to forward the call to an on-site security personnel.

Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. Click **Add** to add an inbound route for PSAP callbacks.
3. In the **Name** field, specify a name to help you identify it.
4. In the **Caller ID Pattern** section, add all the emergency numbers that you have set on the PBX.
    a. Click **Add**.
    b. In the **Pattern** field, enter the emergency number.
    c. **Optional:** To add another emergency number, repeat **step a - b**.
5. In the **Trunk** section, select the trunks that are used for emergency calls to the **Selected** box.
6. In the **Default Destination** field, select **Extension**, and select the user who is responsible for answering the returned calls from PSAP.
7. Leave other fields as the default settings.
8. Click **Save** and **Apply**.

Result

When a PSAP operator calls back, the call will be forwarded to the extension user that is configured on the inbound route.

Related information

Set up Basic Emergency Calling
Set up Enhanced Emergency Calling

Manage Emergency Numbers

After you add emergency numbers, you can edit or delete them.

Edit an emergency number

1. Log in to PBX web portal, go to **Call Control > Emergency Number**, click beside the emergency number that you want to edit.
2. Edit information of emergency number.
3. Click **Save** and **Apply**.

Delete an emergency number

1. Log in to PBX web portal, go to **Call Control > Emergency Number**, click beside the emergency number that you want to delete.
2. In the pop-up dialog box, click **OK** to confirm.
3. Click **Apply**.

Export and Import Emergency Numbers

The emergency numbers configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired emergency numbers in the exported file, and import the file to PBX again. This topic describes how to export and import emergency numbers.

Export emergency numbers

You can export all emergency numbers to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Control > Emergency Number**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Emergency Number Parameters.

Import emergency numbers

We recommend that you export emergency numbers to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Emergency Number Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The emergency numbers in the CSV file will be displayed in the **Emergency Number** list.

Related information

Import and Export -FAQ

Emergency Notification Contacts

Add an Emergency Notification Contact

When a user makes an emergency call, Yeastar P-Series Cloud Edition sends a notification to remind the emergency contacts that who dialed which emergency number.

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number > Notification Contacts**.
2. Click **Add**.
3. In the **Notification Contact** field, select a contact type to receive emergency notifications.
   - **Specific Extension**: Send emergency notification to a specific extension (for example, receptionist).

     If you select this contact type, select a desired extension from the **Specific Extension** drop-down list.

- **The Extension Group Manager of the Extension who dialed the emergency number**: Send emergency notification to the extension group manager of the extension who dialed the emergency number.
- **Specific Group Manager**: Send emergency notification to the manager of a specific extension group.

  If you select this contact type, select the desired extension group from the **Specific Group Manager** drop-down list.
- **Custom**: Send emergency notification to an external contact.

  If you select this contact type, enter a contact name in the **Contact Name** field.

4. In the **Notification Method** field, select a notification method.
   - **Send Email**: The PBX will send notifications to the Email address of the contact.

     For more information about emergency Email template, see Configure Emergency Notification Email.

     > **Note:**
     > - To ensure that PBX can successfully send notifications to the Email address, make sure that the Email Server is configured correctly.
     > - If the notification contact is an extension user, make sure that an effective Email address is associated with the user's extension.

   - **Call Mobile**: The PBX will call the mobile number of the contact, and play an announcement.

     For more information about the announcement, see Configure Emergency Notification Prompt.

     > **Note:**
     > To ensure that PBX can successfully call the mobile number, make sure that the **Prefix** is configured correctly according to the outbound route rule.

   - **Call Extension**: The PBX will call the extension number of the contact, and play an announcement.

     For more information about the announcement, see Configure Emergency Notification Prompt.
5. Click **Save**.

Manage Emergency Notification Contacts

After you add emergency notification contacts, you can edit or delete them.

Edit an emergency notification contact

1. Log in to PBX web portal, go to **Call Control > Emergency Number > Notification Contact**, click  beside the emergency notification contact that you want to edit.
2. Edit emergency notification contact or notification method.
3. Click **Save**.

Delete an emergency notification contact

1. Log in to PBX web portal, go to **Call Control > Emergency Number > Notification Contact**.
2. To delete an emergency notification contact, do as follows:
    a. Click  beside the desired contact.
    b. In the pop-up dialog box, click **OK**.
3. To delete emergency notification contacts in bulk, do as follows:
    a. Select the checkboxes of the desired contacts, click **Delete**.
    b. In the pop-up dialog box, click **OK**.

Configure Emergency Notification Email

Yeastar P-Series Cloud Edition provides a default email template for emergency notification, you can also customize your own template.

Background information

By default, Yeastar P-Series Cloud Edition sends emergency notification emails in the language that you have set in system email template. An emergency notification Email contains the following information:

- **Caller information**: Include extension name and number.
- **Emergency information**: Include emergency name and number, and emergency call time.
- **PBX information**: Include PBX nameand PBX serial number.

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number > Notification Contact**.
2. Click **Email Template**.
3. Customize email template.
    a. In the **Template** drop-down list, select **Custom**.
    b. Edit email subject and content according to your needs.
    c. Click **Save**.

Configure Emergency Notification Prompt

Yeastar P-Series Cloud Edition provides a default voice prompt for emergency call notification, you can also customize your own prompt.

Background information

The default emergency announcement reminds the contacts that who dialed which emergency number.

Procedure

1. Log in to PBX web portal, go to **Call Control > Emergency Number > Notification Contact**.
2. Click **Notification Prompt**.
3. In the pop-up window, change the notification prompt.
    a. In the **Prompt** drop-down list, select a desired prompt or upload a custom prompt.

    > **Note:**
    > The upload prompt file should meet the audio file requirements.

    b. In the **Prompt Repeat Count** field, set how many times to play the prompt.
    c. Click **Save**.

Business Hours and Holidays

Overview of Business Hours and Holidays

This topic describes different types of time defined in the Yeastar P-Series Cloud Edition. Read the concepts before you manage business hours and holidays.

Business hours

Business Hours is the working hours during which you conduct business. A rest break that allows an employee to rest for a short period of time during working days is also considered as Business Hours.

Yeastar P-Series Cloud Edition allows you to set a global business hours and also supports custom business hours for designated users.

- **Global Business Hours**

  Global Business Hours is the main business hours for your company. Global Business Hours may apply to most of the employees who have fixed work schedules.

  For more information, see Set Global Business Hours and Route Inbound Calls based on Global Business Hours.
- **Custom Business Hours**

  Custom Business Hours is typically for departments with different hours from your main business hours. You need to create custom schedules that accommodate each department's unique hours and call handling needs.

  For more information, see Route Inbound Calls based on Department Hours.
- **Custom Time Periods**

  Custom Time Periods is typically for individual employees who have their own work schedules.

  For more information, see Route Inbound Calls based on Employee Hours.

Holidays

Holiday defines the days your business is closed due to a holiday. Holidays can be divided into two types:

- Fixed-date Holidays
- Moveable-date Holidays

You can add holidays by date, month, or week according to the holiday type, and set different prompts for different holidays. For more information, see Create a Holiday.

Outside Business Hours

Outside Business Hours is the time periods that are not defined as Business Hours or Holidays.

Related information

Set Global Business Hours
Monitor Time Condition Status
Create a Holiday
Override Time Condition for Inbound Calls
Automatic Reset of Time Condition

Global Business Hours

Set Global Business Hours

This topic describes how to set up Global Business Hours.

Background information

Global Business Hours is the main business hours for your company. Global Business Hours may apply to most of the employees who have fixed work schedules.

For more information about different types of time in Yeastar P-Series Cloud Edition, see Overview of Business Hours and Holidays.

Procedure

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Business Hours**, click **Add**.
2. In the **Business Hours** section, click **Add**, then specify the hours when your business is open.
3. In the **Break Hours** section, click **Add**, then specify rest breaks during the working days.
4. In the **Date Settings** section, select your working days.
    - **Days of Week**: If enabled, you can only use the **Days of Week** as the date condition for your business hours.

- **Advanced Options**: If enabled, you can configure business hours more flexibly with a mixed condition of Week, Month, and Date.

5. **Optional:** In the **Other Options** section, enter a note in the text field to help you identify the time group.
6. Click **Save** and **Apply**.

Result

- A time group is created for Global Business Hours.
- You can create more time groups according to your company's business hours. All the time groups created on the **Business Hours** page are regarded as your company's global business hours.

What to do next

- To handle inbound calls based on the Global Business Hours, see Route Inbound Calls based on Global Business Hours.
- To limit users to make outbound calls based on the Global Business Hours, see Set up an Outbound Route.

Related information

Monitor Time Condition Status
Override Time Condition for Inbound Calls
Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on Employee Hours

Manage Global Business Hours

This topic describes how to edit and delete the time groups that you've defined as your Global Business Hours.

Edit a time group of Global Business Hours

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays**.
2. On the **Business Hours** page, select a desired time group, click  .
3. Change the time settings.

4. Click **Save** and **Apply**.

   The Global Business Hours is updated.

Delete a time group of Global Business Hours

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays**.
2. On the **Business Hours** page, select a desired time group, click **Delete**.
3. In the pop-up dialog box, click **OK** to confirm.

   The time group is deleted from the Global Business Hours.

Related information

Set Global Business Hours
Overview of Business Hours and Holidays

Holidays

Create a Holiday

This topic describes how to create holidays by date, week, and month.

Create a holiday by date

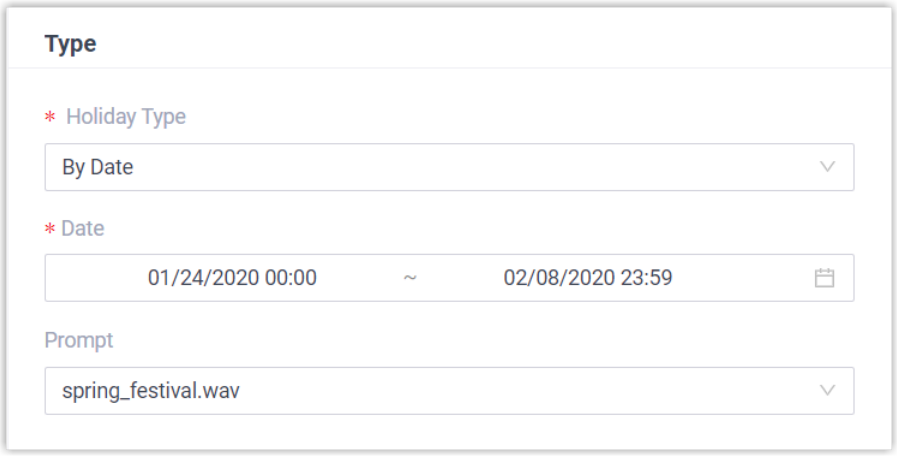
If the holiday date varies every year, you can create a holiday by date.

Example

Chinese Spring Festival varies every year, and 2020 Chinese Spring Festival falls on Jan. 24 to Feb. 8. You can set the holiday as follows.

Configuration Example

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. On the **Holidays** page, click **Add**.
3. In the **Basic** section, enter `2020 Chinese Spring Festival` in the text field.
4. In the **Type** section, set the type, date, and prompt of the holiday.



- **Holiday Type**: Select **By Date**.
- **Date**: Select the holiday start date and end date.
- **Prompt**: Optional. Select an existing prompt or click **Upload** to upload a prompt.

> **Note:**
> - Prompts in the drop-down list are synchronized from **PBX Settings > Voice Prompt > Custom Prompt**.
> - To play the prompt to callers when they call to the PBX during holidays, you need to enable **Play Holiday Prompt During Holidays** (Path: **Call Control > Inbound Route > Default Destination > Play Holiday Prompt During Holidays**) for the time-based inbound route.

5. Click **Save** and **Apply**.

Create a holiday by month

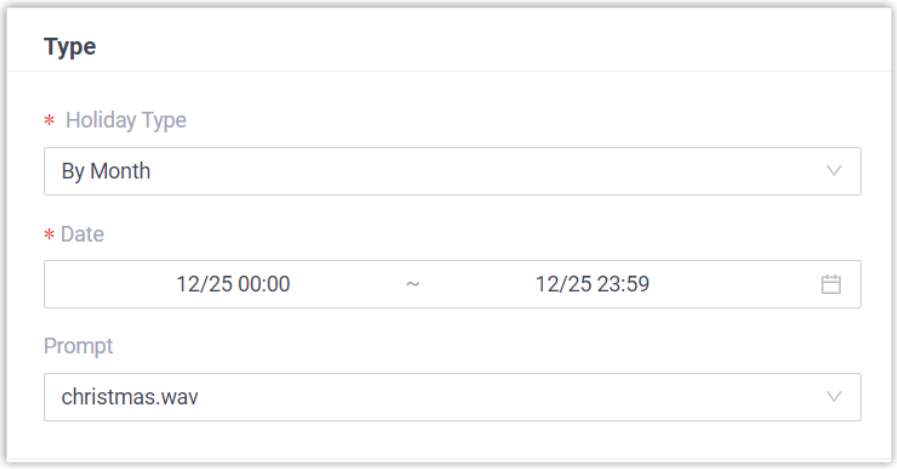
If the holiday always falls on the same date, you can set a holiday by month.

Example

The Christmas falls on Dec. 25 every year. You can set the holiday as follows.

Configuration Example

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. On the **Holidays** page, click **Add**.
3. In the **Basic** section, enter `Christmas` in the text field.
4. In the **Type** section, set the type, date, and prompt of the holiday.



   - **Holiday Type**: Select **By Month**.
   - **Date**: Select the holiday start date and end date.
   - **Prompt**: Optional. Select an existing prompt or click **Upload** to upload a prompt.

   > **Note:**
   > - Prompts in the drop-down list are synchronized from **PBX Settings > Voice Prompt > Custom Prompt**.
   > - To play the prompt to callers when they call to the PBX during holidays, you need to enable **Play Holiday Prompt During Holidays** (Path: **Call Control > Inbound Route > Default Destination > Play Holiday Prompt During Holidays**) for the time-based inbound route.

5. Click **Save** and **Apply**.

Create a holiday by week

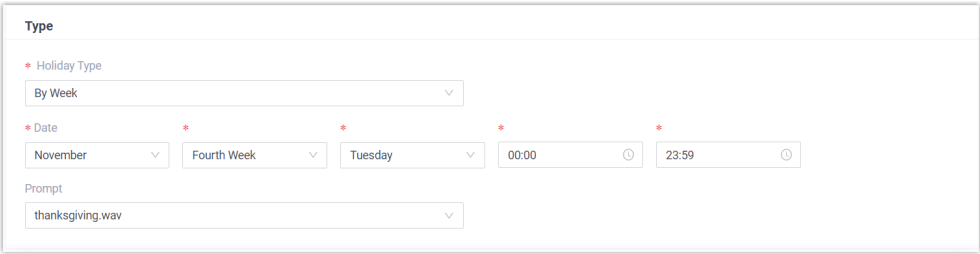
If a holiday always falls on the same week, you can set a holiday by week.

Example

Thanksgiving Day falls on the fourth Thursday of November. You can set the holiday as follows.

Configuration Example

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. On the **Holiday** page, click **Add**.
3. In the **Basic** section, enter `Thanksgiving Day` in the text field.
4. In the **Type** section, set the type, date, and prompt of the holiday.



   - **Holiday Type**: Select **By Week**.
   - **Date**: Select the month and the day of a specific week.
   - **Prompt**: Optional. Select an existing prompt or click **Upload** to upload a prompt.

   > **Note:**
   > - Prompts in the drop-down list are synchronized from **PBX Settings > Voice Prompt > Custom Prompt**.
   > - To play the prompt to callers when they call to the PBX during holidays, you need to enable **Play Holiday Prompt During Holidays** (Path: **Call Control > Inbound Route > Default Destination > Play Holiday Prompt During Holidays**) for the time-based inbound route.

5. Click **Save** and **Apply**.

What to do next

- To handle inbound calls based on Holidays, see Set up an Inbound Route.
- To limit users to make outbound calls during holidays, see Set up an Outbound Route.

Related information

Overview of Business Hours and Holidays
Manage Holidays
Export and Import Holidays

Manage Holidays

This topic describes how to edit and delete a holiday.

Edit a holiday

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. On the **Holidays** page, select a desired holiday, click  .
3. Change the holiday settings.
4. Click **Save** and **Apply**.

   The holiday list is updated.

Delete a holiday

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. On the **Holidays** page, select a desired holiday, click **Delete**.
3. In the pop-up dialog box, click **OK** to confirm.

   The holiday is deleted from the holiday list.

Related information

Create a Holiday
Overview of Business Hours and Holidays
Export and Import Holidays

Export and Import Holidays

The holidays configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired holidays in the exported file, and import the file to PBX again. This topic describes how to export and import holidays.

Export holidays

You can export all holidays to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Holidays Parameters.

Import holidays

We recommend that you export holidays data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Holidays Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Control > Business Hours and Holidays > Holidays**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The holidays in the CSV file will be displayed in the **Holidays** list.

Related information

Overview of Business Hours and Holidays
Create a Holiday
Manage Holidays

Time Condition

Time Condition Overview

Time Condition is the communication feature in Yeastar P-Series Cloud Edition that enables you to set up distinct time periods for call handling. Time Condition allows you to route calls to various destinations at a different time like business hours, outside business hours, and holidays.

Where can Time Condition be applied?

Time Condition can be applied to an Inbound Route, an Outbound Route, and extension presence switch.

Apply to an Inbound Route

Time Condition can be used to control the destination of an inbound call based on date and time.

When a call reaches PBX, PBX will check the current system date and time against the time group associated, and then route the call to corresponding destination.



Apply to an Outbound Route

Time Condition can be used to limit the use of an Outbound Route based on date and time.

When a call is made, the system will check the current system date and time against the time group associated. Only when the time comes to the permitted time group can the outbound call be made.

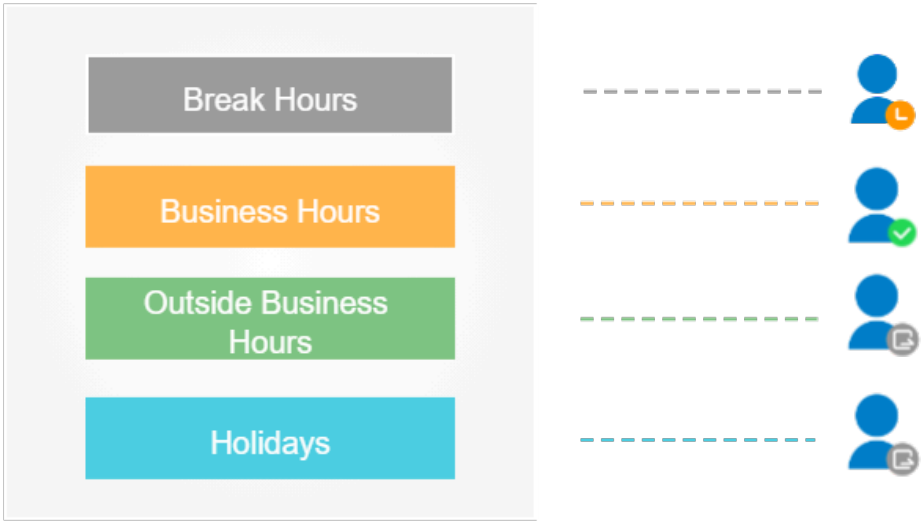


Apply to extension presence switch

Extension presence status can be auto switched based on Time Condition. When time condition changes, extension presence status would be changed accordingly.

For example, when the PBX is in the time condition for Business Hours, the extension status is auto switched to **Available**.

> **Note:**
> This feature should be enabled on extension configuration page. For more information, see Automatically Switch Extension Presence Based on Time.



Time Condition override

For a time-based inbound route, the system routes inbound calls based on the system time. However, users may need to force open or close business occasionally. In this case, you can grant permissions to allow specific users to override time condition for inbound calls through the route. For more information, see Override Time Condition for Inbound Calls.

Note:

- If users do not manually clear time condition override, the system will automatically reset the time condition. For more information, see Automatic Reset of Time Condition.
- To keep the time condition after overriding, see Disable automatic reset of time condition.

Related information

Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on Employee Hours
Set up an Outbound Route

Allow Users to Override Time Condition by Feature Code

By default, all the users can NOT override time condition. To allow users to override time condition by feature code, follow instructions in the topic to grant permissions to specific users.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. Scroll down the page, and configure the permission of time condition override in the **Switch Business Hours and Holidays Status** section.

   Table 15.

Setting	Feature Code	Permission
Switch Global Business Hours and Holidays Status (for Global Business Hours)	*99	Select desired extensions.
Time Condition Switching Prefix (for Custom Business Hours and Custom Time Periods)	Starting with prefix *8 **Note:** Feature codes starting with *8 would be generated for inbound routes that are based on Custom Business Hours or Custom Time Periods.	

3. Click **Save** and **Apply**.

Related information

Override Time Condition for Inbound Calls
Monitor Time Condition Status
Automatic Reset of Time Condition
Enable or Disable Automatic Reset of Time Condition

Allow Users to Override Time Condition on Operator Panel

By default, all the users can NOT override time condition. To allow users to override time condition on Operator Panel, follow instructions in the topic to grant permissions to specific users.

Restrictions

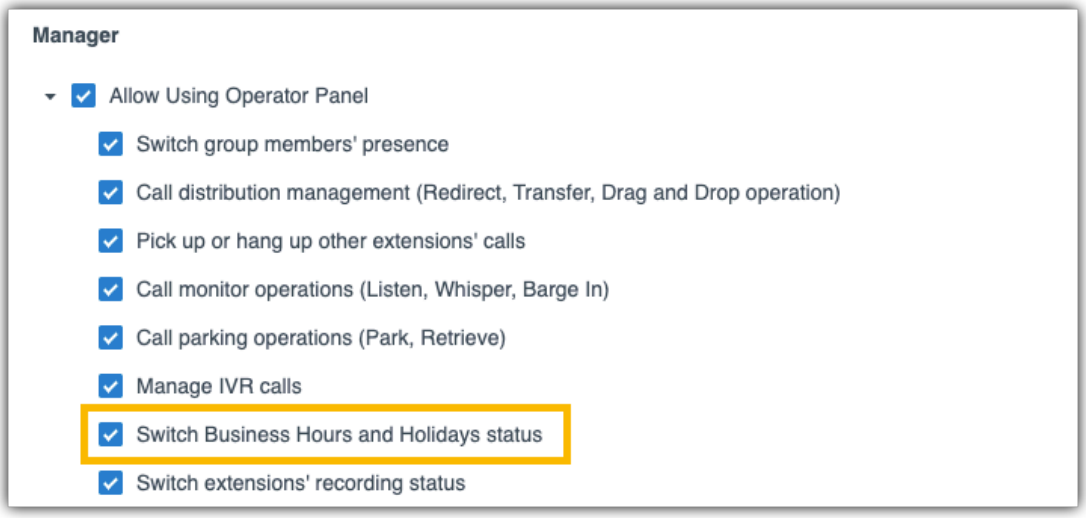
On Operator Panel, only time condition for Global Business Hours can be overridden.

Procedure

The permission of time condition override can only be assigned to extension group managers. Follow the instructions below to grant permission to extension group managers.

> **Note:**
> If you want to grant permissions to a specific user, you can assign a custom user type to the member, and customize permissions. For more information, see Assign a custom user type to a group member.

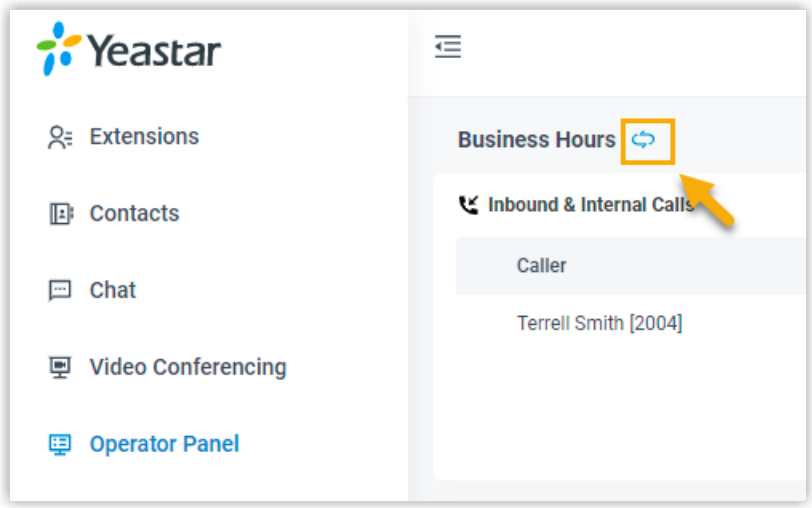
1. Log in to PBX web portal, go to **Extension and Trunk > Extension Group**.
2. Select an extension group, and click .
3. On the **Extension Group** page, click **Group Permissions** tab.
4. In the **Permission Configuration** section, select the checkbox of **Switch Business Hours and Holidays status**.



5. Click **Save** and **Apply**.

Result

On Operator Panel, the user whose user type is Manager can click to override time condition for Global Business Hours.



For more information, see Override Time Condition on Operator Panel.

Related information

Allow Users to Override Time Condition by Feature Code
Automatic Reset of Time Condition
Enable or Disable Automatic Reset of Time Condition

Override Time Condition for Inbound Calls

If you have configured a time-based inbound route, the system will automatically route calls to different destinations based on time. However, users may need to force open or close business occasionally. This topic describes how to achieve time condition override for inbound calls.

Background information

Users may need to override time condition in the following scenarios:

- **Temporary night shift**

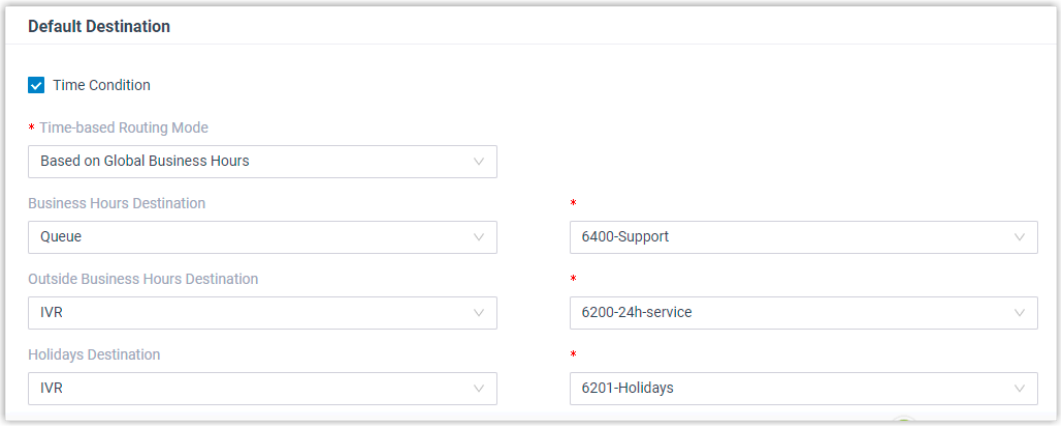
  After business hours, the employee who needs to work in the night can force open the business hours to provide communication services for customers.
- **Occasionally leaving**

  Your company may close the business earlier than usual on a special day. For example, your company will close the business one hour in advance on the Christmas day and you can force close business before you leave.

Override time condition for inbound calls (Global Business Hours)

Background information

An inbound route based on Global Business Hours is set up as follows:



Procedure

Follow the instructions below to achieve time condition override for inbound calls through the route:

1. Grant permission to allow specific users to override time condition.
2. To override time condition, the authorized users should dial corresponding feature code (default ***99**).

   Inbound calls would be routed to different destinations based on the time when users dial feature code.

   Table 16.

Operate Time	Result
Dial *99 during **Business Hours**	Inbound calls will be routed to **Outside Business Hours** destination (IVR 6200).
Dial *99 during **Outside Business Hours**	Inbound calls will be routed to **Business Hours** destination (Queue 6400).
Dial *99 during **Holidays**	Inbound calls will be routed to **Business Hours** destination (Queue 6400).

3. To clear time condition override, the authorized users should dial feature code (default: *99) again.

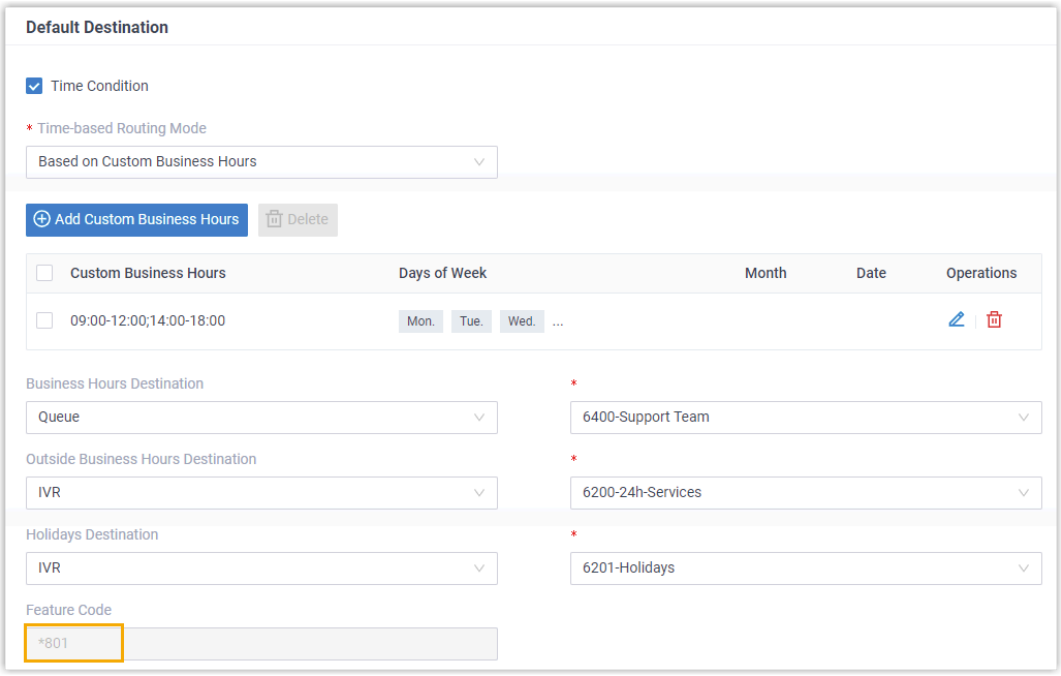
Note:

- If users do not manually clear time condition override, the system will automatically reset the time condition. For more information, see Automatic Reset of Time Condition.
- To keep the time condition after overriding, see Disable automatic reset of time condition.

Override time condition for inbound calls (Custom Business Hours)

Background information

An inbound route based on Custom Business Hours is set up as follows:



Procedure

Follow the instructions below to achieve time condition override for inbound calls through the route:

1. Grant permission to allow specific users to override time condition.
2. To override time condition, the authorized users should dial feature code *801.

   Inbound calls would be routed to different destinations based on the time when users dial feature code.

Table 17.

Operate Time	Result
Dial *801 during **Business Hours**	Inbound calls will be routed to **Outside Business Hours** destination (IVR 6200).
Dial *801 during **Outside Business Hours**	Inbound calls will be routed to **Business Hours** destination (Queue 6400).
Dial *801 during **Holidays**	Inbound calls will be routed to **Business Hours** destination (Queue 6400).

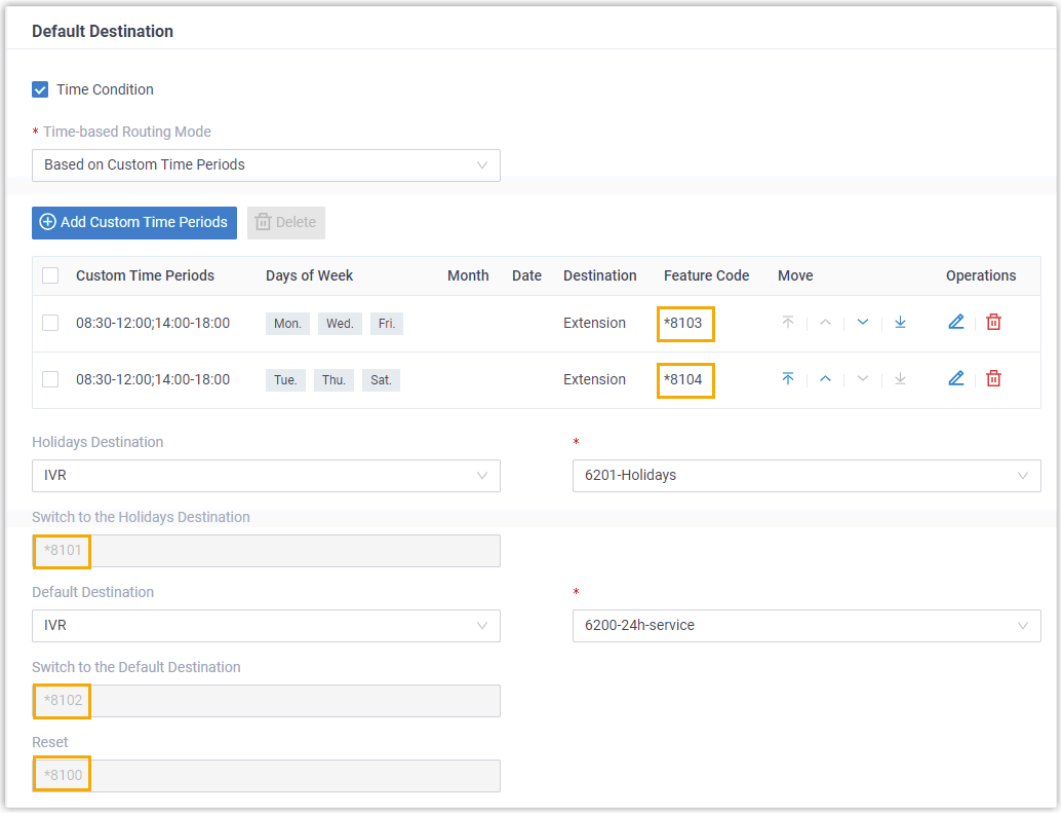
3. To clear time condition override, the authorized users should dial *801 again.

> **Note:**
> - If users do not manually clear time condition override, the system will automatically reset the time condition. For more information, see Automatic Reset of Time Condition.
> - To keep the time condition after overriding, see Disable automatic reset of time condition.

Override time condition for inbound calls (Custom Time Periods)

Background information

An inbound route based on Custom Time Periods is set up as follows:



Procedure

Follow the instructions below to achieve time condition override for inbound calls through the route:

1. Grant permission to allow specific users to override time condition.
2. To override time condition, the authorized users should dial a specific feature code.

   Inbound calls would be routed to corresponding destination based on the dialed feature code.

Table 18.

Time Group	Description	Destination	Feature Code
Time Period 1	Monday, Wednesday, Friday • 08:30 - 12:00 • 14:00 - 18:00	Extension (2000-Leo Ball)	*8103
Time Period 2	Tuesday, Thursday, Saturday	Extension	*8104

Time Group	Description	Destination	Feature Code
	• 08:30 - 12:00 • 14:00 - 18:00	(2004-Terrell Smith)	
Holidays		IVR (6201-Holidays)	*8101
Outside Business Hours		IVR (6200-24h-Services)	*8102

3. To clear time condition override, the authorized users should dial the **Reset** feature code *8100.

> **Note:**
> - If users do not manually clear time condition override, the system will automatically reset the time condition. For more information, see Automatic Reset of Time Condition.
> - To keep the time condition after overriding, see Disable automatic reset of time condition.

Related information

Allow Users to Override Time Condition by Feature Code
Allow Users to Override Time Condition on Operator Panel
Monitor Time Condition Status
Automatic Reset of Time Condition
Enable or Disable Automatic Reset of Time Condition

Monitor Time Condition Status

This topic describes how to set BLF keys on IP phones to monitor time condition status. In this way, users can know which time period the system is working and where inbound calls would be routed.

Background information

Users can monitor time condition status in the following ways:

Monitor time condition status on IP phone

For the users who want to monitor time condition status on their phones, you can set a BLF key for each user.

For more information, see the followings:

- Monitor time condition status for inbound calls (Global Business Hours)
- Monitor time condition status for inbound calls (Custom Business Hours)
- Monitor time condition status for inbound calls (Custom Time Periods)

Monitor time condition status on Operator Panel

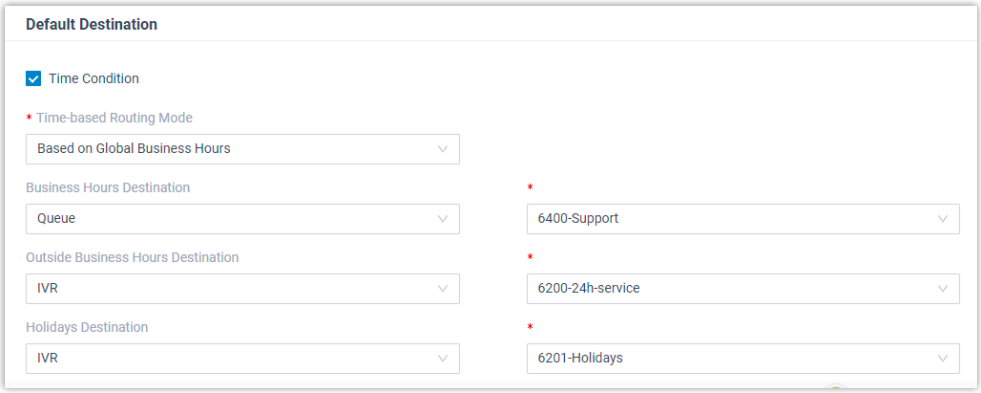
For the users who have permission to access Operator Panel, they can monitor time condition status on Operator Panel directly.

For more information, see Monitor Time Condition Status on Operator Panel.

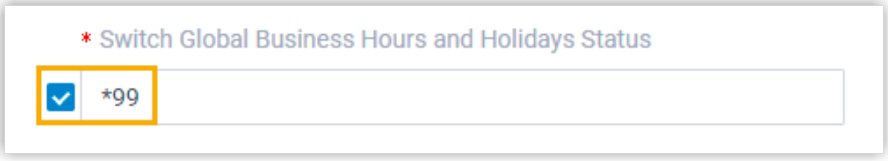
Monitor time condition status for inbound calls (Global Business Hours)

Background information

- An inbound route based on Global Business Hours is set up as follows:



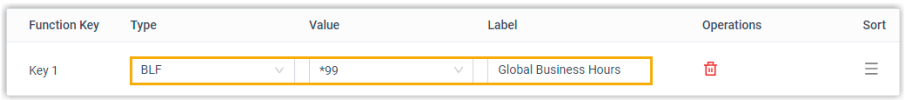
- The feature code for **Switch Global Business Hours and Holidays Status** is *99.



Procedure

1. Assign function keys for extension users to monitor time condition status.
    a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click beside the desired extension.
    b. Click the **Function Keys** tab.
    c. Configure function keys.

    > **Note:**
    > The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

    - **Type**: Select **BLF**.
    - **Value**: Enter the feature code of **Switch Global Business Hours and Holidays Status**. In this example, enter `*99`.
    - **Label**: Optional. Enter a value, which will be displayed on the phone screen.

    d. Click **Save**.
2. If the extension has been provisioned and associated with a phone, reprovision the phone to take effect.
    a. Go to **Auto Provisioning > Phones**.
    b. Click beside the phone assigned to this extension.
    c. In the pop-up window, click **OK** to reprovision the phone.
3. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.

Result

The function key settings are automatically updated on the phone and different BLF LED indicates different status.

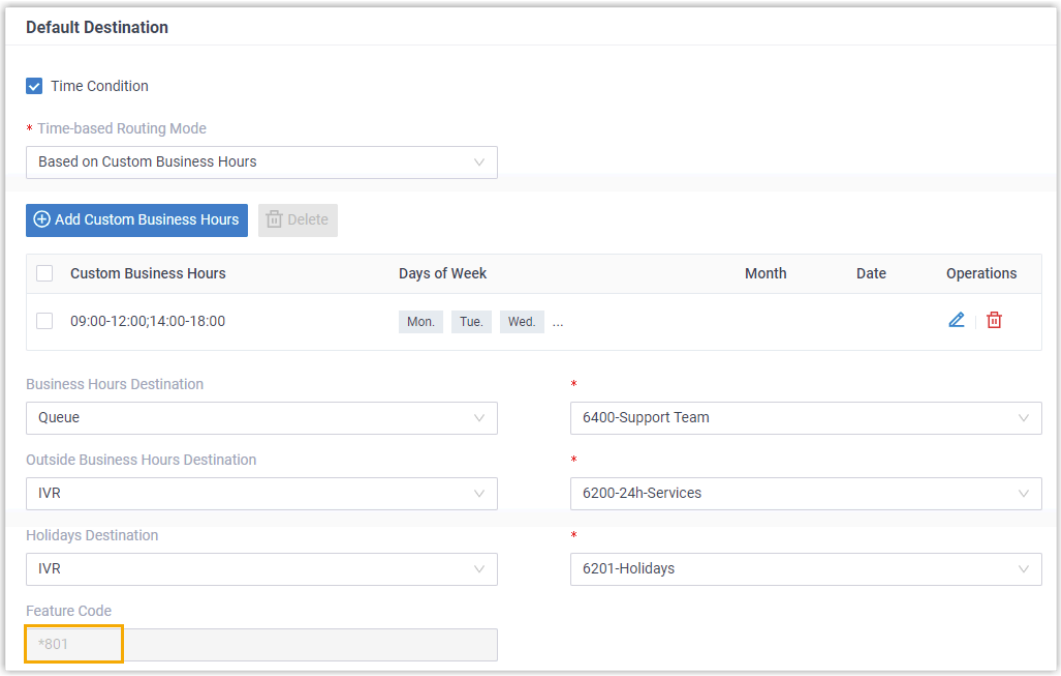
- **Red**: The system is in the status of **Business Hours**.
- **Green**: The system is in the status of **Outside Business Hours** or **Holidays**.

- **Off**: The BLF configurations are incorrect.

Monitor time condition status for inbound calls (Custom Business Hours)

Background information

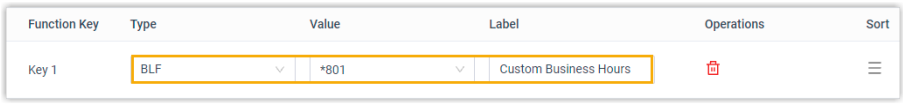
An inbound route based on Custom Business Hours is set up as follows:



Procedure

1. Assign function keys for extension users to monitor time condition status.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click beside the desired extension.
   b. Click the **Function Keys** tab.
   c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.



- **Type**: Select **BLF**.
- **Value**: Enter feature code of the inbound route. In this example, enter `*801`.
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension has been provisioned and associated with a phone, reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.
   c. In the pop-up window, click **OK** to reprovision the phone.
3. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.

Result

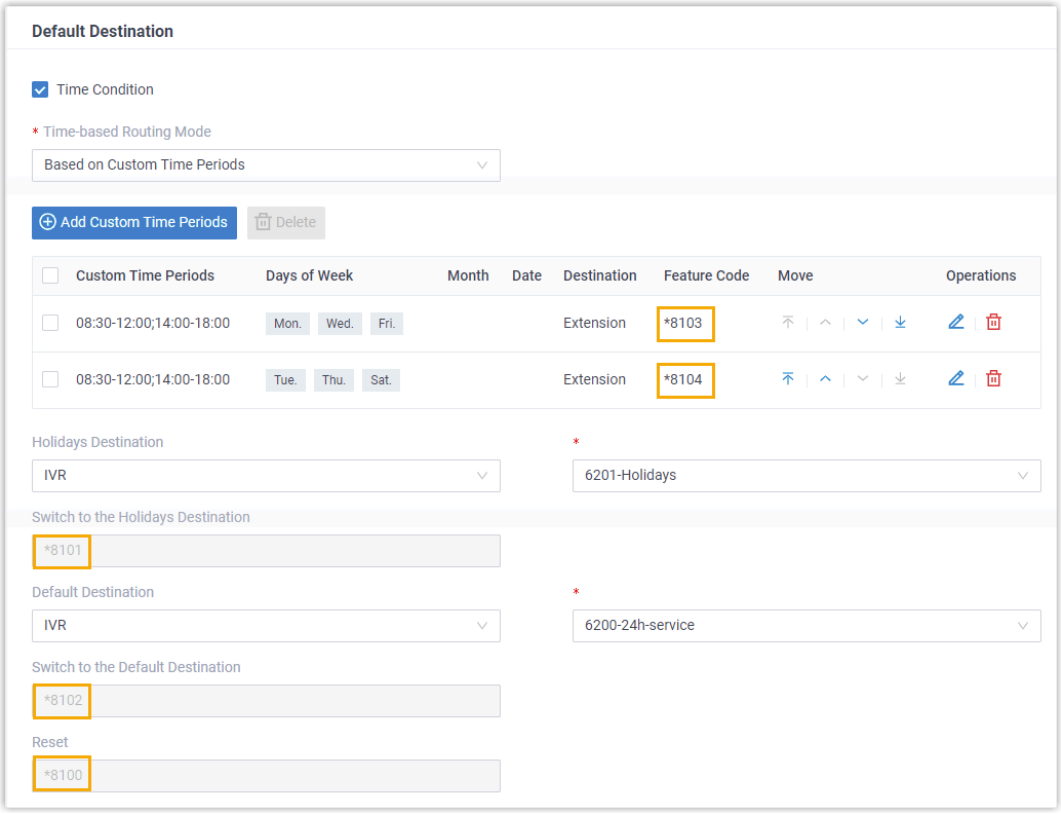
The function key settings are automatically updated on the phone and different BLF LED indicates different status.

- **Red**: The system is in the status of **Business Hours**.
- **Green**: The system is in the status of **Outside Business Hours** or **Holidays**.
- **Off**: The BLF configurations are incorrect.

Monitor time condition status for inbound calls (Custom Time Periods)

Background information

An inbound route based on Custom Time Periods is set up as follows:



Procedure

1. Assign function keys for extension users to monitor time condition status.
    a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, click beside the desired extension.
    b. Click the **Function Keys** tab.
    c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.



- **Type**: Select **BLF**.
- **Value**: Enter the feature codes as needed.
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension has been provisioned and associated with a phone, re-provision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.
   c. In the pop-up window, click **OK** to reprovision the phone.
3. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.

Result

The function key settings are automatically updated on the phone and different BLF LED indicates different status.

- **Red**: Current inbound calls would be routed to the destination.
- **Green**: Current inbound calls would NOT be routed to the destination.
- **Off**: The BLF configurations are incorrect.

> **Note:**
> If the BLF key is set for **Reset** feature code, the BLF LED should be off.

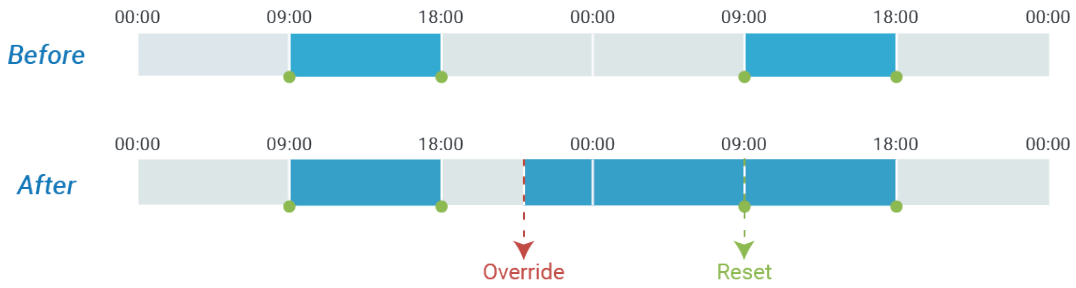
Automatic Reset of Time Condition

By default, if users have overridden time condition and don't clear the time condition override manually, PBX will automatically reset time condition in next period. The next period can be the starting point of business hours, outside business hours, or holidays. This topic gives examples to help you understand how the system automatically resets time condition.

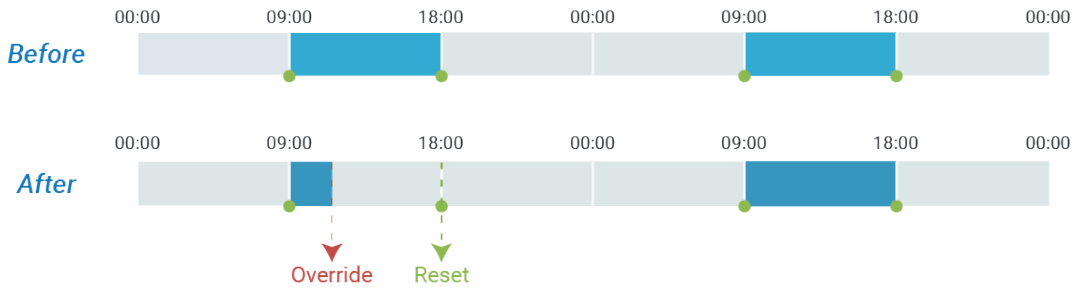
Examples for Global Business Hours/Custom Business Hours

	Business Hours
	Outside Business Hours
	Next hop of time condition: According to the time condition you configured, the system automatically switches the destination for incoming calls at the time point.
	Override time condition
	Automatically reset time condition

Example 1: Override to Business Hours



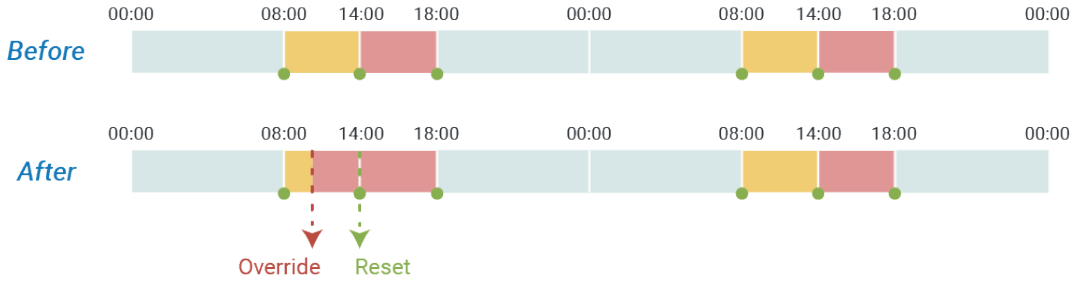
Example 2: Override to Outside Business Hours



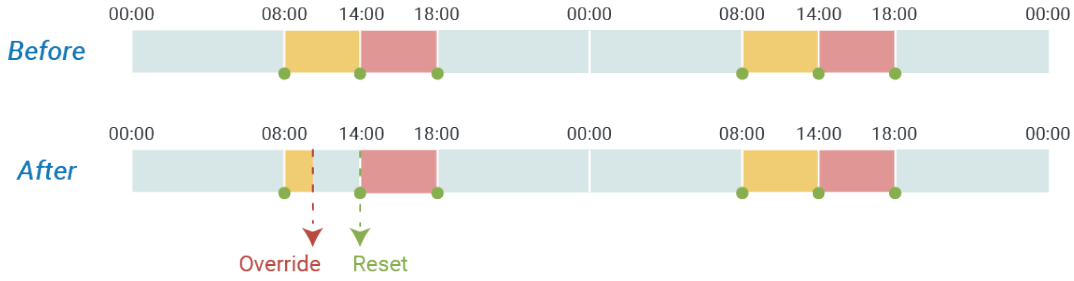
Examples for Custom Time Periods

	Time Period 1
	Time Period 2
	Outside Business Hours
	Next hop of time condition: According to the time condition you configured, the system automatically switches the destination for incoming calls at the time point.
	Override time condition
	Automatically reset time condition

Example 1: Override to Time Period 2



Example 2: Override to Outside Business Hours

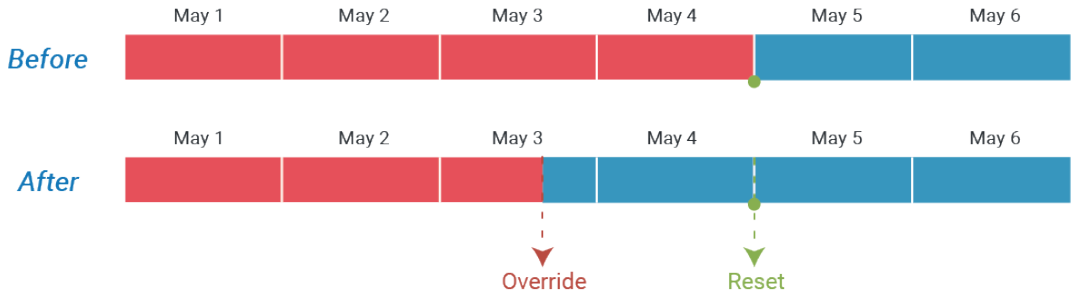


Example for Holidays

	Business Hours

	Holidays
	Next hop of time condition (holiday or non-holiday): According to the time condition you configured, the system automatically switches the destination for incoming calls at the time point.
	Override time condition
	Automatically reset time condition

Example: Override to non-holiday



Related information

Enable or Disable Automatic Reset of Time Condition
Monitor Time Condition Status
Override Time Condition for Inbound Calls

Enable or Disable Automatic Reset of Time Condition

By default, if users have overridden time condition and don't clear the time condition override manually, PBX will automatically reset time condition in next period. You can decide whether to auto reset time condition or keep the time condition after overriding.

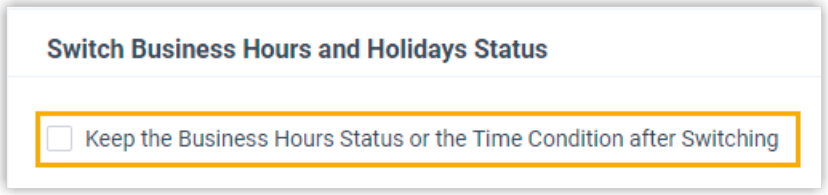
Enable automatic reset of time condition

To make the system go back to the normal schedule after users override time condition, follow the instructions below.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. Scroll down to **Switch Business Hours and Holidays Status** section.

3. Unselect the checkbox of **Keep the Business Hours Status or the Time Condition after Switching**.



4. Click **Save** and **Apply**.

Result

When it comes to the next starting point of business hours, outside business hours, or holidays, the system will reset time condition and go back to the normal schedule.

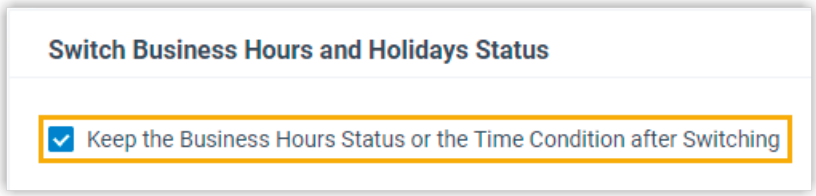
For more information, see Automatic Reset of Time Condition.

Disable automatic reset of time condition

To make the system keep the status after users override time condition, follow the instructions below.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. Scroll down to **Switch Business Hours and Holidays Status** section.
3. Select the checkbox of **Keep the Business Hours Status or the Time Condition after Switching**.



4. Click **Save** and **Apply**.

Result

After users override time condition, the system will stay in the state until someone manually clear the override.

Tip:

Tips for clearing the override:

- For inbound route based on **Global Business Hours**, dial the feature code of **Switch Global Business Hours and Holidays Status**.
- For inbound route based on **Custom Business Hours**, dial the feature code of corresponding inbound route.
- For inbound route based on **Custom Time Periods**, dial the **Reset** feature code.

Related information

Override Time Condition for Inbound Calls
Monitor Time Condition Status

Inbound Route

Inbound Route Overview

An inbound route allows external callers to reach your system and routes the inbound calls to a specific destination based on the pre-configured rules and criteria.

Types of inbound call routing

Yeastar P-Series Cloud Edition has the following types of inbound call routing based on different criteria, such as time, DID numbers, and Caller IDs.

Note:

- If you don't specify any criteria on an inbound route, there will be no restriction on the inbound route. The system will route all inbound calls to the inbound route destination.
- You can set up multiple criteria on an inbound route. For example, route inbound calls based on time and DID number, or route inbound calls based on DID number and Caller ID number.

Time-based call routing

Time-based call routing connects callers to a destination based on the time that they call. The inbound calls are handled differently according to your company's time schedules.

For more information, see the following topics:

- Route Inbound Calls based on Global Business Hours
- Route Inbound Calls based on Department Hours
- Route Inbound Calls based on Employee Hours

DID-based call routing

DID-based call routing connects callers to a destination based on the phone numbers (also known as DID) that the callers dial. Only when the dialed DID numbers match the DID rules on the inbound route will the calls be routed to the destination.

For more information, see Route Inbound Calls based on DID Numbers.

Caller-ID-based call routing

Caller-ID-based call routing allows you to accept or reject calls based on the caller's phone number. Inbound calls that match the Caller ID pattern on PBX will be routed to the pre-configured destination. For those unmatched, calls can not be established.

For more information, see Route Inbound Calls based on Caller ID and Route Inbound Calls by Matched Phonebook Contacts.

Inbound route destinations

Yeastar P-Series Cloud Edition provides various inbound destinations to meet your business needs.

The following options are available to help you decide the inbound route destinations:

- **Extension**
- **Extension Voicemail**
- **Group Voicemail**
- **Match Selected Extensions**
- **DID Range to Extension Range**

- **DID Pattern to Selected Extensions**
- **IVR**
- **Ring Group**
- **Queue**
- **Conference**
- **External Number**
- **Outbound Route**
- **Fax to Email**
- **Hang up**
- **Play Greeting then Hang up**

Set up an Inbound Route

To receive inbound calls from external users, you need to set up at least one inbound route.

Background information

Yeastar P-Series Cloud Edition has a default inbound route that will route all the inbound calls to an IVR. You can delete the default inbound route, and add a new one to configure settings according to your needs.

Prerequisites

Ensure that you have set up at least one trunk for external users to call in.

Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. **Optional:** Set an "alert info text" to add to Alert-info header in INVITE request for inbound calls.

   When receiving an inbound call, the phone will inspect "Alert-Info" header to determine which ring tone it should use for ringing.
4. In the **Trunk** section, select the desired trunks from **Available** box to **Selected** box.

   The PBX will route inbound calls through this inbound route when external users call the selected trunk number.
5. **Optional:** If you want to route inbound calls based on DID numbers, configure **DID Pattern**.

The PBX will route inbound calls only when the callers dial the matched DID numbers.

> **Note:**
> Leave this field blank to match calls with any or no DID information.

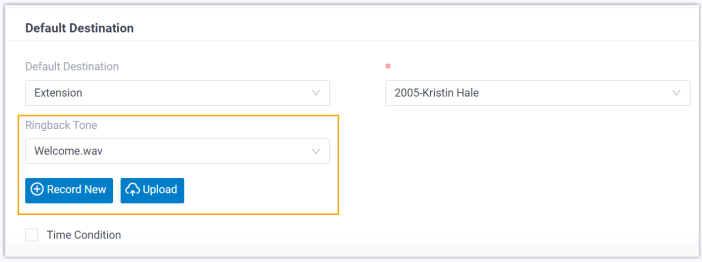
For more information, see Route Inbound Calls based on DID Numbers.

6. **Optional:** If you want to route inbound calls based on Caller IDs, configure **Caller ID Pattern**.

   The PBX will route inbound calls only when the Caller IDs match the Caller ID pattern.

   > **Note:**
   > Leave this field blank to match calls with any or no Caller ID info.

   For more information, see Route Inbound Calls based on Caller ID.
7. Configure the inbound route destination.

   > **Note:**
   > When the destination is set to an extension, you can select or upload a music in the follow-up **Ringback Tone** field, which will be played to the caller before the extension user answers the call.
   >
   > 
   >

   - If you want to route inbound calls to one destination whenever the calls reach the system, perform the following operations:
     a. Keep the **Time Condition** unselected.
     b. Configure the **Default Destination**.
   - If you want to route inbound calls to different destinations based on the time, perform the following operations:
     a. Select the checkbox of **Time Condition**.
     b. Select an option from the drop-down list of **Time-based Routing Mode**.

c. Configure the destinations based on the time.

   If an inbound call reaches the PBX during the time period, PBX will route the call to the selected destination.

d. **Optional:** To prevent inbound calls from being distributed to the holiday destination, select the checkbox of **Ignore the Holiday Destination**.

   Inbound calls during holiday will be distributed to other destinations according to your office hour setting.

e. **Optional:** To play a prompt to callers before routing the inbound calls to the holiday destination, select the checkbox of **Play Holiday Prompt During Holidays**.

> **Note:**
> Make sure that you have set a prompt for the holiday (Path: **Call Control > Business Hours and Holidays > Holidays > Type > Prompt**). Otherwise, the inbound calls will be directly routed to the holiday destination without playing a prompt.

For more information of inbound call routing based on time, see the following topics:

- Route Inbound Calls based on Global Business Hours
- Route Inbound Calls based on Department Hours
- Route Inbound Calls based on Employee Hours

8. **Optional:** To receive faxes through this inbound route, enable **Fax Detection** and configure the fax destination.
   - **Extension**: The faxes will be sent to the selected extension. You need to register the extension on a SIP compatible fax machine.

> **Note:**
> If the selected extension is deleted, the fax destination will automatically jump to **Hang up**, and faxes cannot be received through this inbound route.

   - **Fax to Email**: The faxes will be converted to email attachments and be sent to an extension's email address.

> **Note:**

> Make sure the system email is configured correctly, or **Fax to Email** will fail to work.

For more information of fax setting, see Fax Overview.

9. Click **Save** and **Apply**.

Time Based Inbound Routes

Route Inbound Calls based on Global Business Hours

This topic gives a configuration example to describe how to configure inbound route to control inbound calls based on Global Business Hours, which can be applied to most of the employees.

Background information

Assume that your company's business hours are as follows:

- **Working days**: Monday to Friday
- **Business hours**: 09:00-12:00 and 14:00-18:00

When customers call in the trunk sip_routein_GBH, you want to route the calls based on the time as follows:

- During business hours, route inbound calls to an IVR for business.
- During a holiday, route inbound calls to another IVR for holiday, and play a prompt "office_holiday".
- For other time periods, route inbound calls to a voicemail.

Prerequisites

- The trunk for inbound calling has been set up and is ready for use.
- Global Business Hours is configured according to your company's business hours.
- The desired destination of the inbound route has been configured on the system.

  In this scenario, an IVR for business hours, an IVR for holiday should be preconfigured.

  For more information of IVR, see Set up an IVR.
- If you want to play a prompt to customers when they make calls to the trunk during holidays, a prompt should be preconfigured.

In this scenario, a prompt "office_holiday" is preconfigured.

For more information, see Create a Holiday.

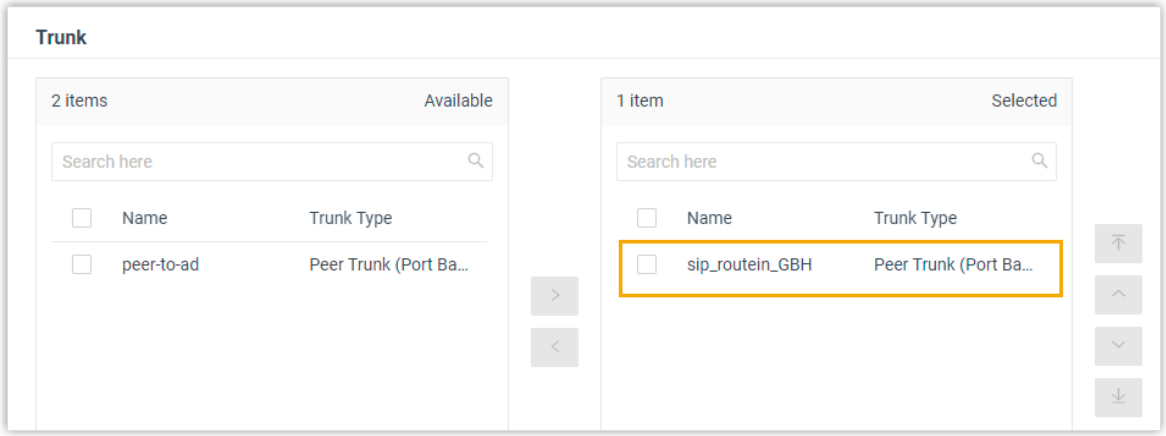
- If you want to set up multiple inbound routes for different time schedules, each inbound route should be associated with a different trunk or a trunk with different DID numbers. In this way, the inbound calls can be always directed to your desired destination.

  How to configure inbound route based on DID numbers, see Route Inbound Calls based on DID Numbers.

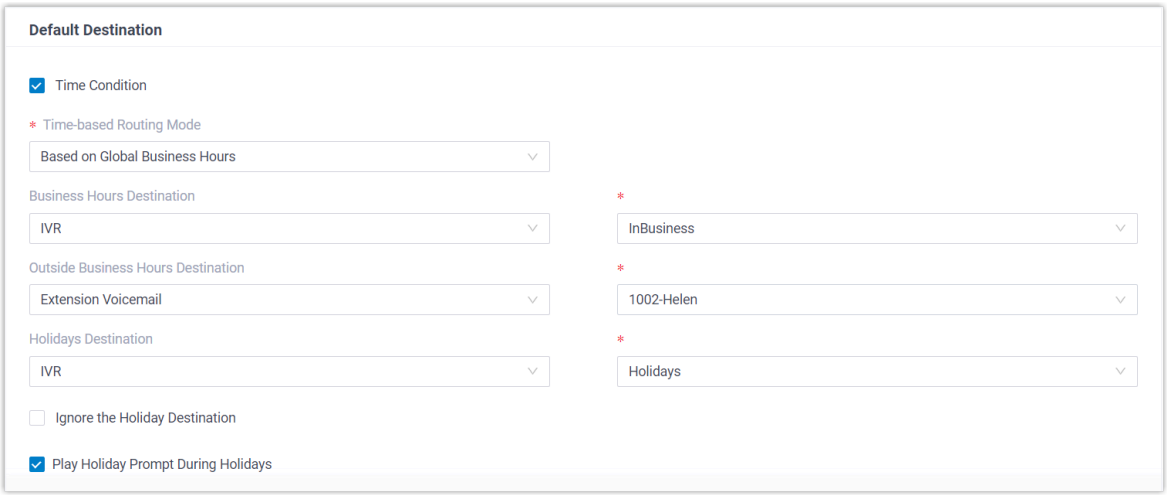
Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Trunk** section, select the desired trunks from **Available** box to **Selected** box.

   In this scenario, select the trunk sip_routein_GBH.



4. In the **Default Destination** section, complete the following operations:



a. Select the checkbox of **Time Condition**.
b. In the drop-down list of **Time-based Routing Mode**, select **Based on Global Business Hours**.
c. Configure the following destinations based on the time.
   - **Business Hours Destination**: Select the destination for inbound calls during global business hours.

     In this scenario, select **IVR**, and select the IVR for business hours.
   - **Outside Business Hours Destination**: Outside Business Hours is the time periods that are not defined as Business Hours or Holidays.

     In this scenario, select **Extension Voicemail** then select an extension number.
   - **Holidays Destination**: Select the destination for inbound calls during holidays.

     In this scenario, select **IVR**, and select the IVR for holidays.
d. Select the checkbox of **Play Holiday Prompt During Holidays**.

   In this scenario, the system will play the prompt "office_holiday" before routing inbound calls to the holiday destination.

5. Click **Save** and **Apply**.

Result

When customers make calls to the phone number of the selected trunk sip_routein_GBH, the calls will be routed to different destinations based on the time.

Related information

Route Inbound Calls based on Department Hours
Route Inbound Calls based on Employee Hours
Route Inbound Calls based on DID Numbers
Route Inbound Calls based on Caller ID
Route Inbound Calls by Matched Phonebook Contacts

Route Inbound Calls based on Department Hours

This topic gives a configuration example to describe how to configure inbound route to control inbound calls for the departments that maintain different hours from the company's global business hours.

Scenarios

The employees in the branch office's support department have different business hours from the head office. The department hours is listed as below:

- **Working days**: Monday to Friday
- **Business hours**: 21:00 - 23:00 and 00:00 - 05:00

When customers call in the trunk sip_routein_DP, you want to route the calls based on the time as follows:

- During business hours, route inbound calls to the support team's queue.
- During a holiday, route inbound calls to another IVR for holiday, and play a prompt "office_holiday".
- For other time periods, route inbound calls to a voicemail.

Prerequisites

- The trunk for inbound calling has been set up and is ready for use.
- The desired destination of the inbound route should be configured on the system.

  In this scenario, a queue and an IVR for holiday should be preconfigured.

  For more information about the configurations of queue and IVR, see Create a Queue and Set up an IVR.
- If you want to play a prompt to customers when they make calls to the trunk during holidays, a prompt should be preconfigured.

  In this scenario, a prompt "office_holiday" is preconfigured.

For more information, see Create a Holiday.

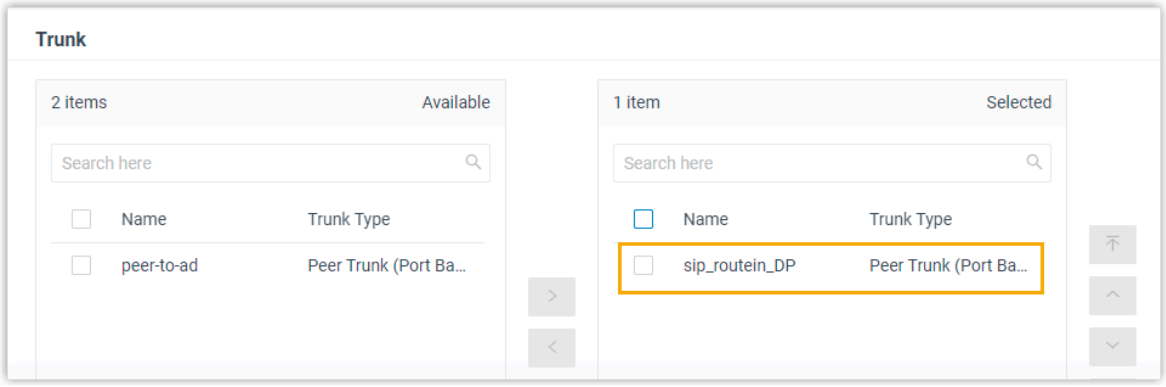
- If you want to set up multiple inbound routes for different time schedules, each inbound route should be associated with a different trunk or a trunk with different DID numbers. In this way, the inbound calls can be always directed to your desired destination.

  How to configure inbound route based on DID numbers, see Route Inbound Calls based on DID Numbers.

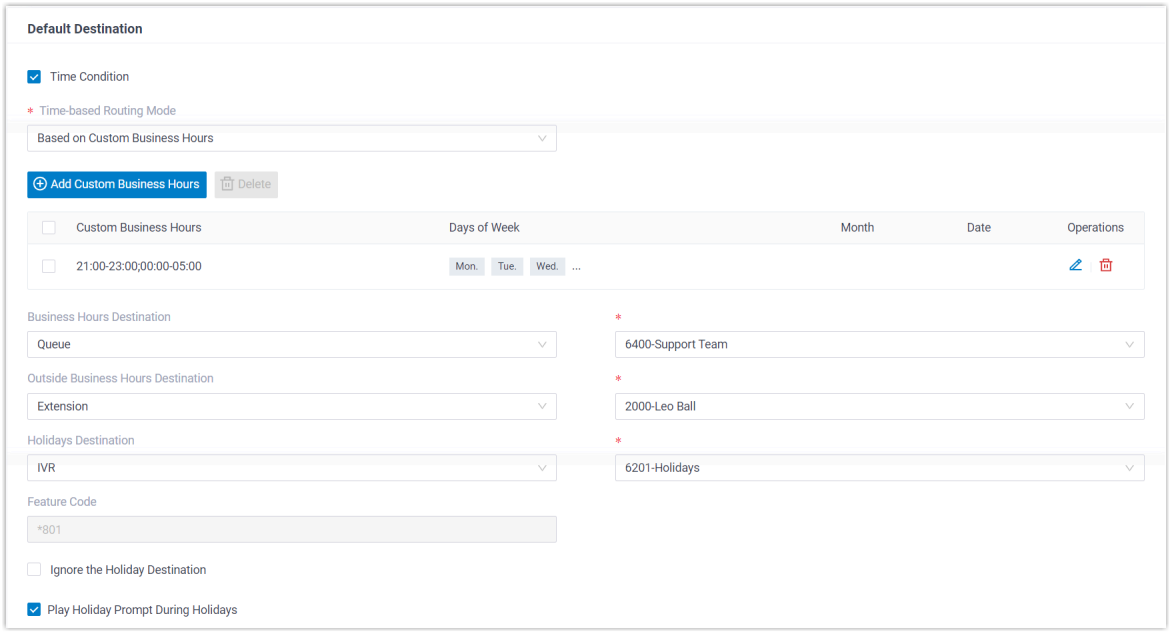
Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Trunk** section, select the desired trunks from **Available** box to **Selected** box.

   In this scenario, select the trunk sip_routein_DP.



4. In the **Default Destination** section, complete the following operations:



a. Select the checkbox of **Time Condition**.
b. In the drop-down list of **Time-based Routing Mode**, select **Based on Custom Business Hours**.
c. Configure custom business hours.
    i. Click **Add Custom Business Hours**.
    ii. In the **Custom Business Hours** section, click **Add** to add business hours.

    In this scenario, add two business hours, 21:00 - 23:00 and 00:00 - 05:00.
    iii. In the **Date Settings** section, select working days.

    In this scenario, select **Days of Week** and select days from Monday to Friday.
    iv. Click **Confirm**.
d. Configure the following destinations based on the time.
    - **Business Hours Destination**: Select the destination for inbound calls during business hours.

      In this scenario, select **Queue**, and select the Queue "Support Team".
    - **Outside Business Hours Destination**: Outside Business Hours is the time periods that are not defined as Business Hours or Holidays.

      In this scenario, select **Extension Voicemail**, and select an extension.
    - **Holidays Destination**: Select the destination for inbound calls during holidays.

      In this scenario, select **IVR**, and select the IVR for holidays.

e. Select the checkbox of **Play Holiday Prompt During Holidays**.

In this scenario, the system will play the prompt "office_holiday" before routing inbound calls to the holiday destination.

5. Click **Save** and **Apply**.

Result

- When customers make calls to the phone number of the selected trunk sip_routein_-DP, the calls will be routed to different destinations based on time.
- Feature code ***801** is generated for the inbound route. The authorized user can dial *801 to override time condition of the inbound route. For more information, see Override Time Condition for Inbound Calls.

Related information

Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Employee Hours
Route Inbound Calls based on DID Numbers
Route Inbound Calls based on Caller ID
Route Inbound Calls by Matched Phonebook Contacts

Route Inbound Calls based on Employee Hours

This topic gives a configuration example to describe how to configure inbound route to control inbound calls for individual employees who have their own work schedules.

Scenarios

Duty doctors in a hospital are responsible for supporting emergency patient needs or arranging appointments for patients over phone calls.

- Each duty doctor has a different time schedule and will provide services based on the time schedule.
- During the time periods that no doctors are on duty or when it comes to a holiday, the incoming calls from patients will be routed to an IVR.

The following shows time schedule for the duty doctors.

Doctor Name	Time Schedule
Dr. Tommy Tse	Monday 07:00 -12:00

Doctor Name	Time Schedule
	Friday 12:00 - 18:00
Dr. Eric Chan	Monday 00:00 - 07:00 Thursday 07:00 - 12:00

Prerequisites

- The trunk for inbound calling has been set up and is ready for use.
- The desired destination of the inbound route should be configured on the system.

  In this scenario, an IVR should be configured to ensure that patients can reach their desired services.

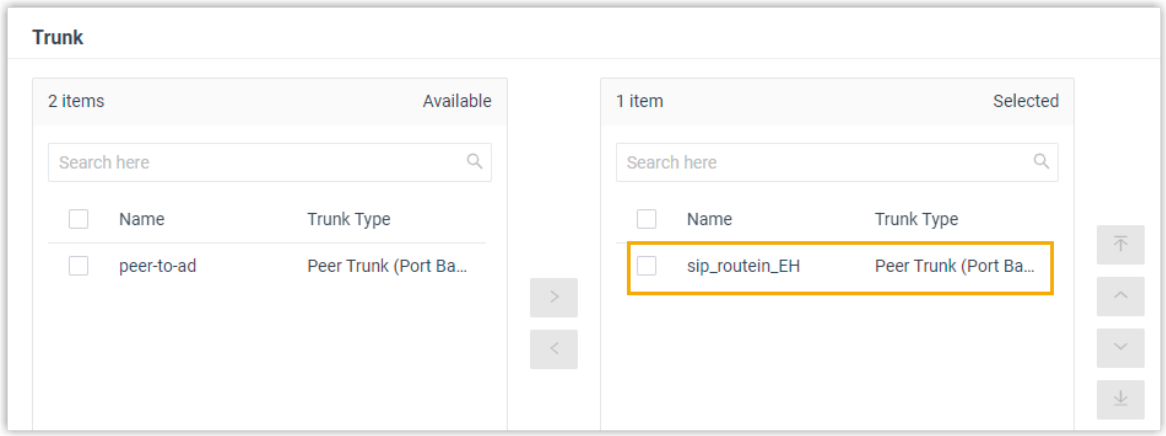
  For more information about IVR, see Set up an IVR.
- If you want to set up multiple inbound routes for different time schedules, each inbound route should be associated with a different trunk or a trunk with different DID numbers. In this way, the inbound calls can be always directed to your desired destination.

  How to configure inbound route based on DID numbers, see Route Inbound Calls based on DID Numbers.

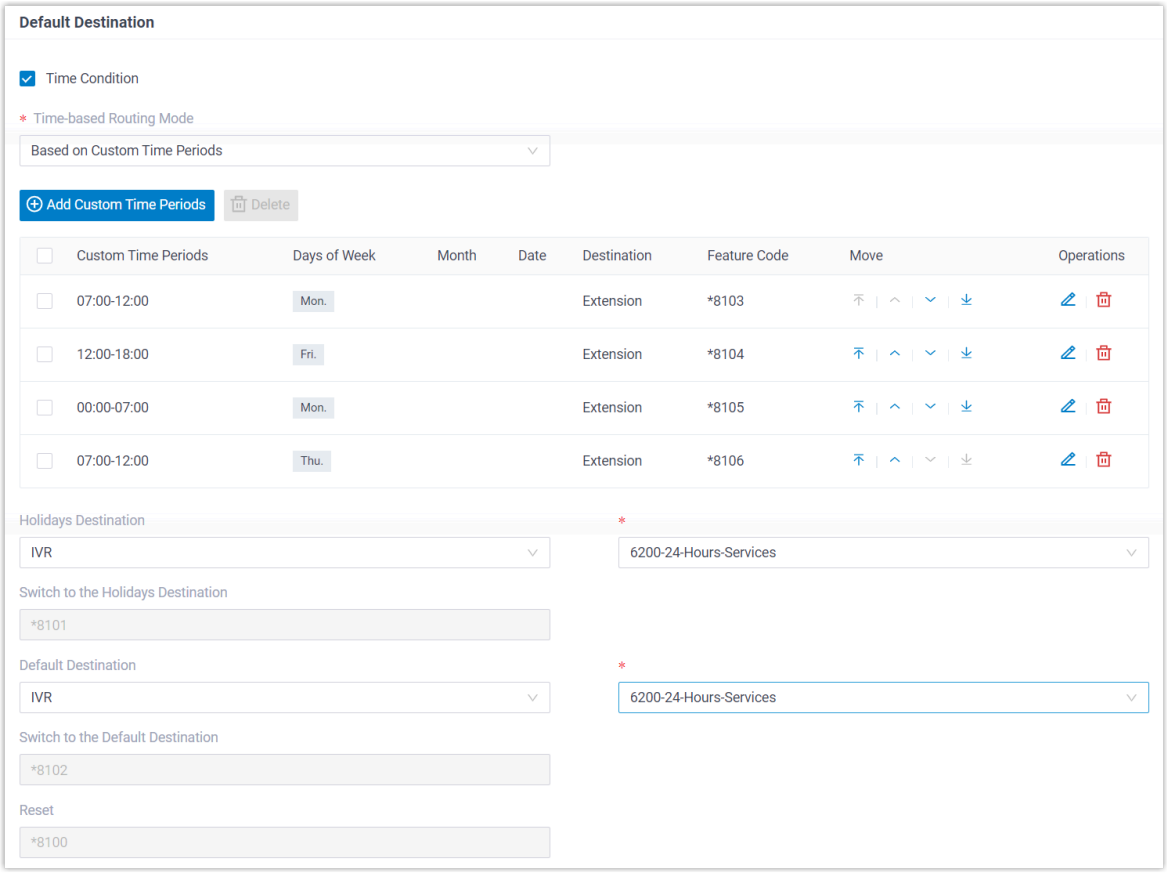
Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Trunk** section, select the desired trunks from **Available** box to **Selected** box.

   In this scenario, select the trunk sip_routein_EH.



4. In the **Default Destination** section, complete the following operations:



a. Select the checkbox of **Time Condition**.
b. In the drop-down list of **Time-based Routing Mode**, select **Based on Custom Time Periods**.
c. Add time schedule for the duty doctors.
    i. Click **Add Custom Time Periods**.

ii. In the pop-up window, click **Add** to add time periods, set relevant destinations, and select days of week.
iii. Click **Confirm**.
iv. Repeat **step i - iii** to add another time schedule.

In this scenario, add four time schedules as below.

Start Time	End Time	Days of Week	Destination
07:00	12:00	Monday	Tommy's extension
12:00	18:00	Friday	Tommy's extension
00:00	07:00	Monday	Eric's extension
07:00	12:00	Thursday	Eric's extension

d. Configure the **Holidays Destination**.

In this scenario, select **IVR**, and select an IVR to guide patients.

e. Configure the **Default Destination**.

In this scenario, select **IVR**, and select an IVR to guide patients.

5. Click **Save** and **Apply**.

Result

- When external users make calls to the selected trunk sip_routein_EH, the calls will be routed to different destinations based on time:
  - During the custom time periods, inbound calls go to the specified destination.
  - During the rest of time that is not defined, inbound calls go to the **Default Destination**.
  - When it comes to holiday, inbound calls go to the **Holidays Destination**.
- Feature codes are generated as follows.

> **Tip:**
> The authorized users can dial a specific feature code to override time condition and route inbound calls to corresponding destinations. For more information, see Override Time Condition for Inbound Calls.

  - One feature code for each time period.

- A **Switch to the Holidays Destination** feature code that allows for switching destination of inbound calls to **Holidays**.
- A **Reset** feature code that allows for removing any overrides currently set.
- A **Switch to the Default Destination** feature code that allows for switching destination of inbound calls to the default destination.

Related information

Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on DID Numbers
Route Inbound Calls based on Caller ID
Route Inbound Calls by Matched Phonebook Contacts

Caller ID/DID Based Inbound Routes

Route Inbound Calls based on DID Numbers

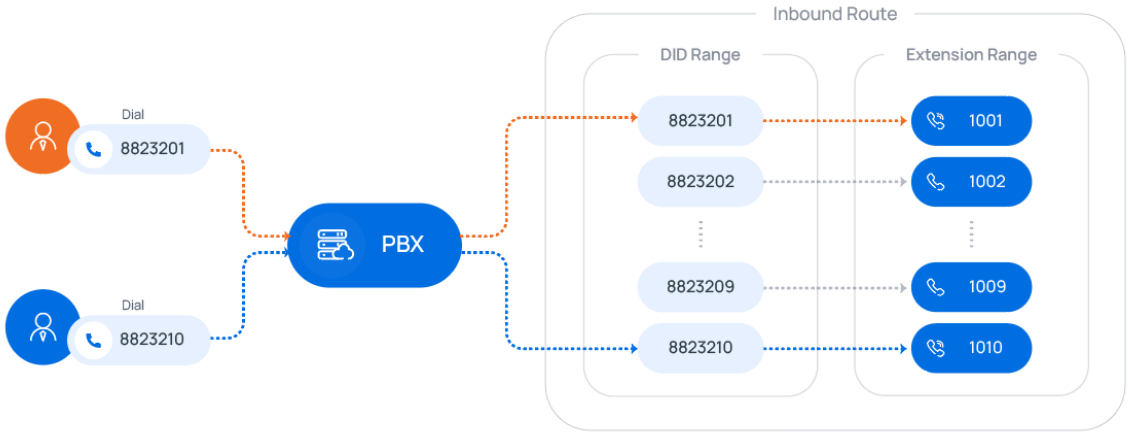
This topic gives configuration examples to describe how to route inbound calls based on the dialed numbers (also called DID numbers).

DID routing modes

Yeastar P-Series Cloud Edition provides four DID matching modes to help you route inbound calls based on DID numbers.

- **Match DID Range to Extension Range**

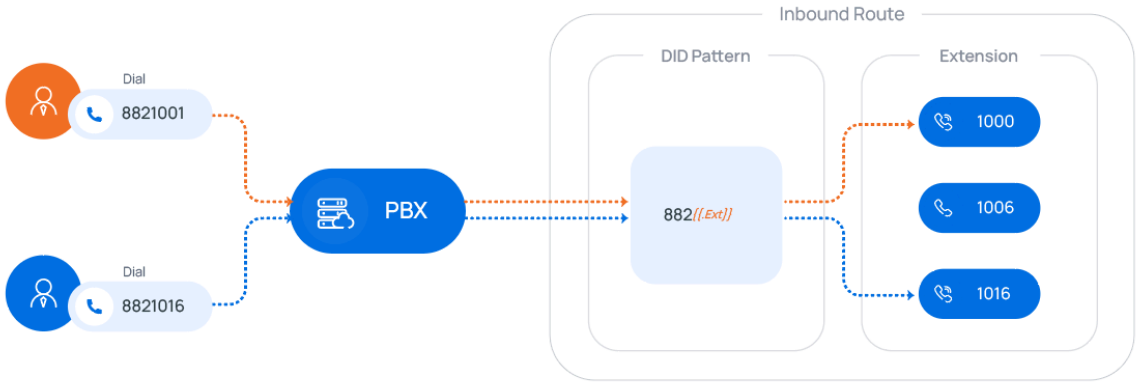
  Match DID Range and Extension Range in a one-to-one correspondence with sequential order.



For more information, see the configuration example Route calls to extension users by matching DID range.

- **Match DID Pattern to Extensions**

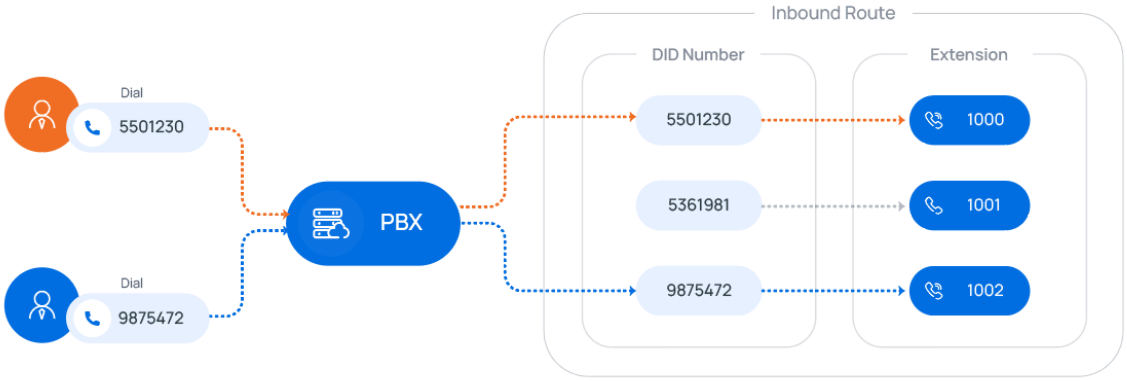
  Use the variable `{{.Ext}}` to match extension number in the DID pattern.



For more information, see the configuration example Route calls to extension users by matching DID patterns.

- **DID Number to Specific Extension**

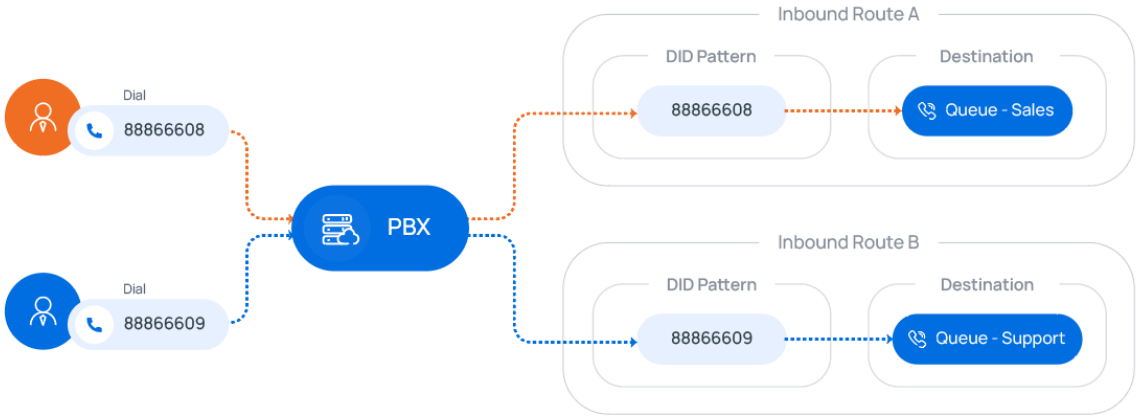
  The inbound calls that match the defined DID number(s) will be routed to the specified extension(s) in one-to-one correspondence.



For more information, see the configuration example Route calls to extension users by matching specific DID numbers.

- **DID Pattern**

  The inbound calls that match the defined DID number(s) or pattern(s) will be routed to a defined destination.



For more information, see the configuration example Route calls to different destinations by matching DID patterns.

Route calls to extension users by matching DID range

Scenario

Company ABC purchases a SIP trunk, and gets 10 DID numbers that are in order: 8823201-8823210.

The company wants to redirect inbound calls to specific extensions based on the provided DID numbers as follows:

Table 19.

DID Number	Extension Number
8823201	1001
8823202	1002
8823203	1003
8823204	1004
8823205	1005
8823206	1006
8823207	1007
8823208	1008
8823209	1009
8823210	1010

Prerequisites

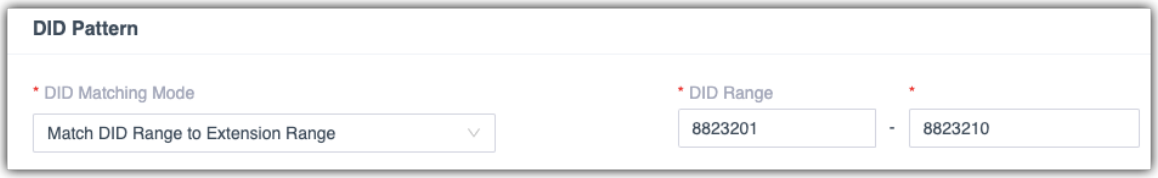
- You have purchased DID numbers from the trunk provider.
- The trunk for inbound calling has been set up and is ready for use.

Configuration example

According to this scenario, configure an inbound route based on the DID ranges as follows:

- **Name**: Enter a name to help you identify it.
- **Trunk**: Select the trunk that binds the DID numbers.
- **DID Pattern**:
  - **DID Matching Mode**: Select **Match DID Range to Extension Range**.
  - **DID Range**: Enter the start number and the end number of the DID range.

    In this scenario, enter *8823201* and *8823210*.



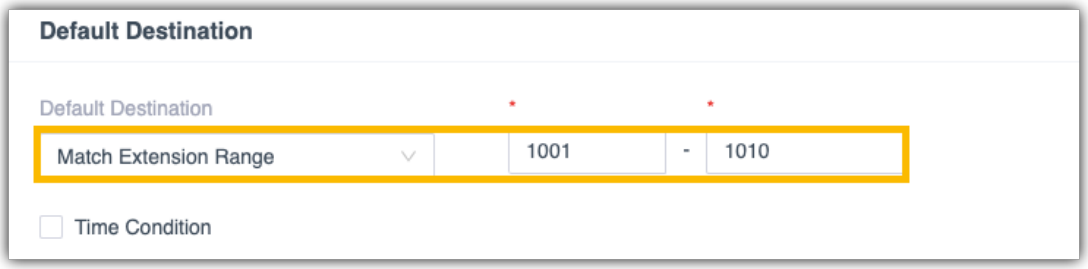
- **Caller ID Pattern**: Leave it blank, which means no limit on the inbound caller ID.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

In this scenario, route inbound calls to the default destination whenever the calls reach the system.

- **Default Destination**: Select **Match Extension Range**, and enter the extension range *1001 - 1010*.

> **Note:**
> The DID range and extension range should have the same size (for example, DID range 991000-991003 and the extension range 1000-1003 have the same size).

- **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

When an external user dials a number that is in the DID range, the user can reach a corresponding extension user directly.

For example, if an external user dials 8823201, the call goes to the extension 1001 directly.

Route calls to extension users by matching DID patterns

Scenario

Company ABC purchases a SIP trunk, and gets 3 DID numbers as follows.

- 8821001, 8821006, 8821016

The company wants to redirect inbound calls to specific extensions based on the provided DID numbers as follows:

> **Note:**

The provided DIDs have the following characteristics:

- Not consecutive
- Each DID number consists of a string of fixed digits and a specific extension number.

Table 20.

DID Number	Extension Number
8821001	1001
8821006	1006
8821016	1016

Prerequisites

- You have purchased DID numbers from the trunk provider.
- The trunk for inbound calling has been set up and is ready for use.

Configuration example

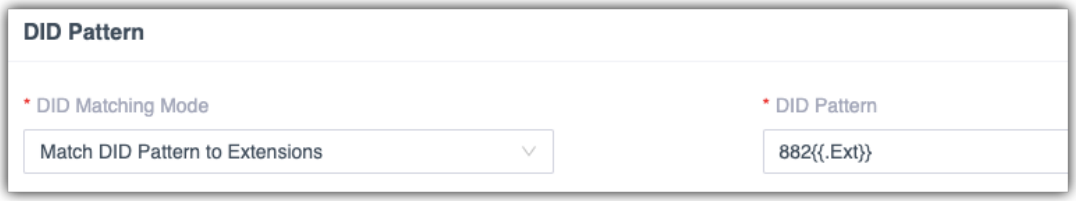
According to this scenario, configure an inbound route based on DID patterns as follows:

- **Name**: Enter a name to help you identify it.
- **Trunk**: Select the trunk that binds the DID numbers.

  In this scenario, select **siptrunk**.
- **DID Pattern**:
  - **DID Matching Mode**: Select **Match DID Pattern to Extensions**.
  - **DID Pattern**: Enter the DID pattern according to the provided DIDs.

    In this scenario, enter *882{{.Ext}}*.

    Note:
    - *{{.Ext}}* is a variable that will match the destination extension.
    - The wildcard `.` and `!` are not allowed.
    - Only one DID pattern is allowed.

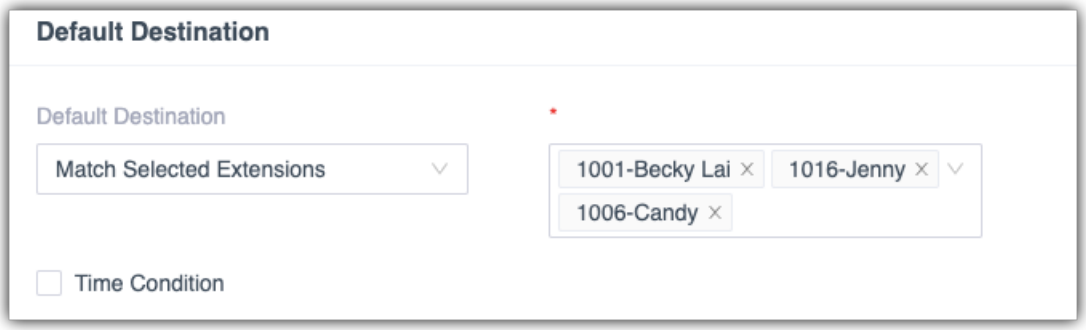


- **Caller ID Pattern**: Leave it blank, which means no limit of inbound caller ID.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this scenario, route inbound calls to the default destination whenever the calls reach the system.

  - **Default Destination**: Select **Match Selected Extensions**, and select extensions.

    In this scenario, select extension 1001, 1006, and 1016.
  - **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

When an external user dials a number that matches the DID pattern, the user can reach a specific extension user directly.

For example, if the external user dials 8821001, the call goes to the extension 1001 directly.

Route calls to extension users by matching specific DID numbers

Scenario

Company ABC purchases a SIP trunk, and gets 3 DID numbers as follows.

- 5501230, 5361981, 9875472

The company wants to redirect inbound calls to specific extensions based on the provided DID numbers as follows:

Table 21.

DID Number	Extension Number
5501230	1000
5361981	1001
9875472	1002

Prerequisites

- You have purchased DID numbers from the trunk provider.
- The trunk for inbound calling has been set up and is ready for use.

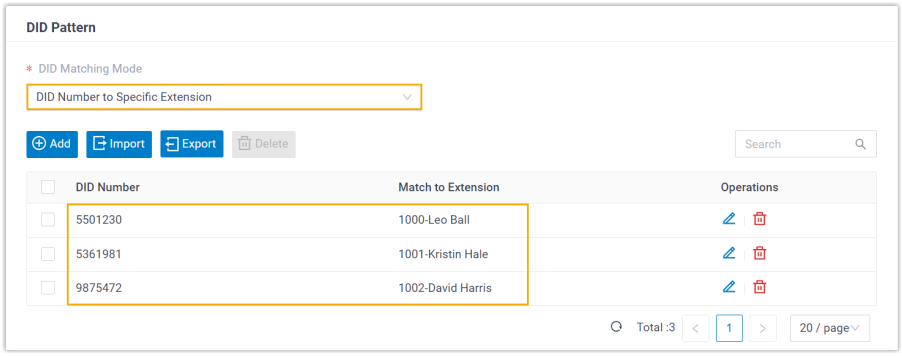
Configuration example

According to this scenario, configure an inbound route based on the specific DID numbers as follows:

- **Name**: Enter a name to help you identify it.
- **Trunk**: Select the trunk that binds the DID numbers.
- **DID Pattern**:
  - **DID Matching Mode**: Select **DID Number to Specific Extension**.
  - **DID Number**: Click **Add** and set up the DID number matching rules.

> **Tip:**
> You can also click **Export** to export the DID number matching rule to a CSV file first, use the file as a template to add DID number matching rules in bulk, then import the file into the inbound route. Refer to DID Number to Specific Extension Parameters to edit the parameters in the CSV file.

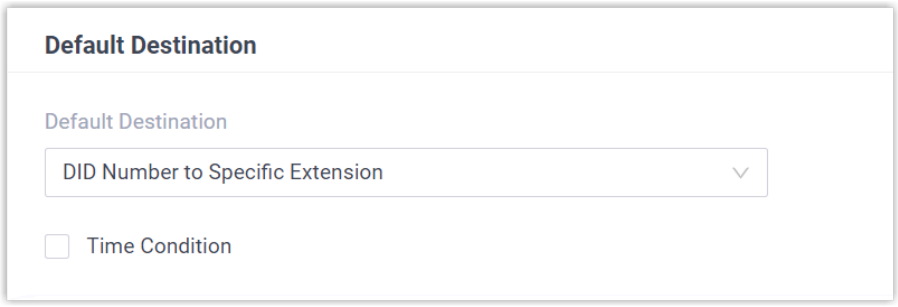
In this example, the configuration is as follows:



- **Caller ID Pattern**: Leave it blank, which means no limit on the inbound caller ID.
- **Default Destination**: Decide whether to route the inbound calls to different destinations based on time and configure the destinations.

  In this example, the configuration is as follows:

  - **Default Destination**: Select **DID Number to Specific Extension**.
  - **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

When an external user dials a number that equals to the DID number, the user can reach a specific extension user directly.

For example, if the external user dials 9875472, the call goes to the extension 1002 directly.

Route calls to different destinations by matching DID patterns

Scenario

Company ABC purchases a SIP trunk, and gets 2 DID numbers as follows.

- 88866608
- 88866609

The company wants to assign the two DID numbers to support team and sales team.

- When external users call 88866609, the calls go directly to support team.
- When external users call 88866608, the calls go directly to sales team.

Prerequisites

- You have purchased DID numbers from the trunk provider.
- The trunk for inbound calling has been set up and is ready for use.

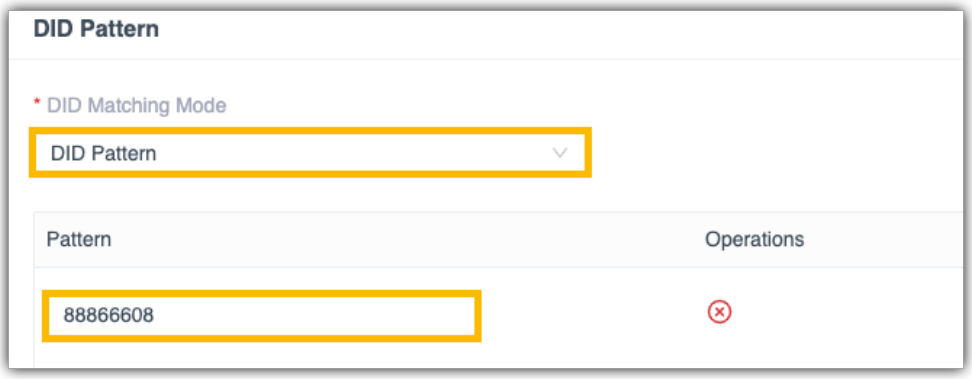
Configuration example

Set up two inbound routes to route calls to different destinations based on DID numbers.

Inbound Route for sales team

- **Name**: Enter a name to help you identify it.
- **Trunk**: Select the trunk that binds the DID numbers.
- **DID Pattern**:
  - **DID Matching Mode**: Select **DID Pattern**.
  - **DID Patterns**: Click **Add** and enter a DID pattern or a DID number.

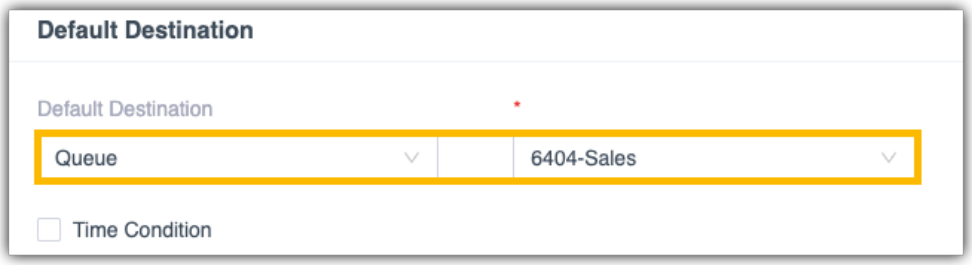
    In this scenario, enter *88866608*.



- **Caller ID Pattern**: Leave it blank, which means no limit of inbound caller ID.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

- **Default Destination**: Select the destination to the queue of sales team.
- **Time Condition**: Unselected.

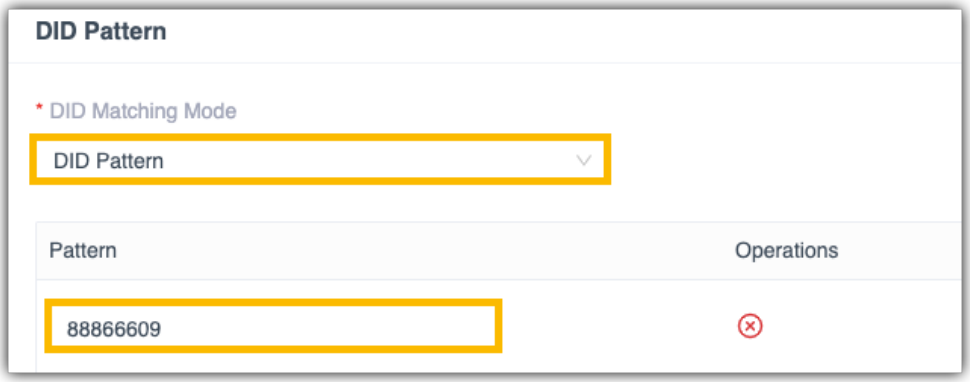


- **Fax Detection**: Leave the settings as default.

Inbound Route for support team

- **Name**: Enter a name to help you identify it.
- **Trunk**: Select the trunk that binds the DID numbers.
- **DID Pattern**:
  - **DID Matching Mode**: Select **DID Pattern**.
  - **DID Patterns**: Click **Add** and enter a DID pattern or a DID number.

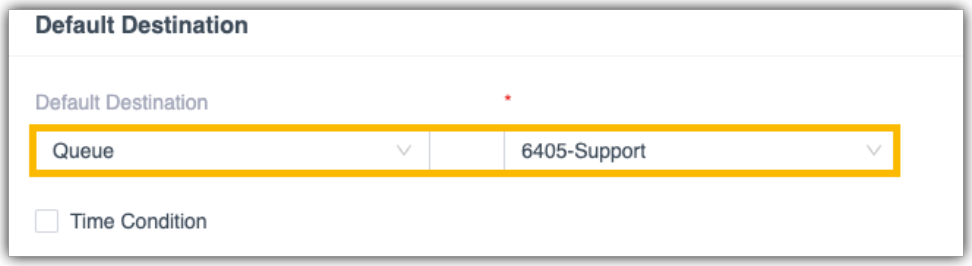
    In this scenario, enter *88866609*.



- **Caller ID Pattern**: Leave it blank, which means no limit of inbound caller ID.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

- **Default Destination**: Select the destination to the queue of support team.
- **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

External users will reach different teams according to the DID numbers they dial.

Related information

Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on Employee Hours
Route Inbound Calls based on Caller ID
Route Inbound Calls by Matched Phonebook Contacts

Route Inbound Calls based on Caller ID

Caller ID routing connects external callers with the appropriate party quickly. This topic gives a configuration example to describe how to route calls by a caller-ID-based inbound route.

Scenarios

Company ABC is a Chinese company that provides consulting services around multiple cities.

For better customer experience, the company has a countrywide toll-free number 400-661-8815 and has multiple teams to provide professional services for customers from different regions.

For example, the following two teams will handle inbound calls based on different caller IDs.

Table 22.

Team	Responsible Region	Area Code
Team-A	Fujian	• 0591 • 0592 • 0593 • 0594 • 0595 • 0596 • 0597 • 0598 • 0599
Team-B	Guangdong	• 0662 • 0663 • 0668 • 0660

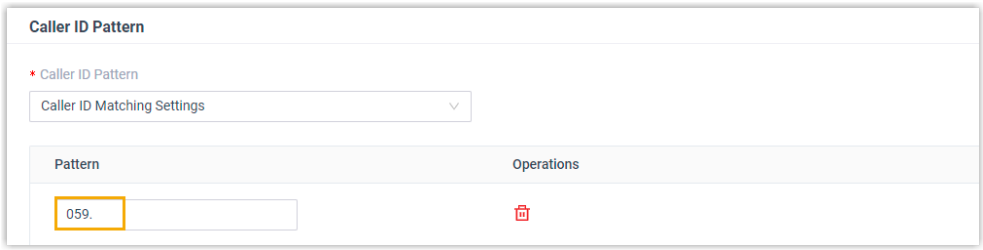
Configuration Example

Set up two inbound routes to route calls to different destinations based on caller IDs.

Inbound Route for Team-A

- **Name**: Enter a name to help you identify it.
- **DID Pattern**: Leave it blank, which means no limit of DID numbers.
- **Caller ID Pattern**: Select **Caller ID Matching Settings**, click **Add** and enter a Caller ID pattern or a full Caller ID.

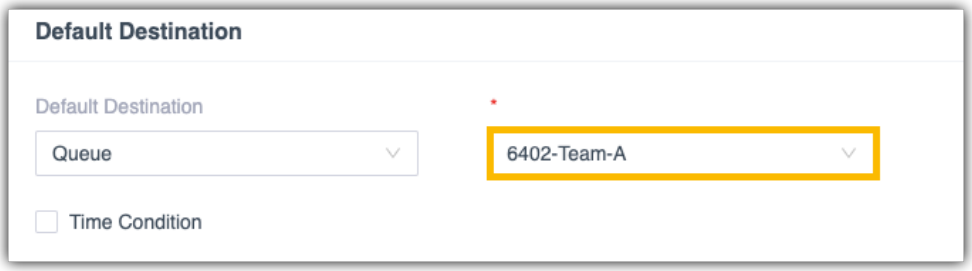
  In this scenario, enter `059.`, which matches all inbound caller IDs that start with digit 059. For more information of Caller ID pattern, see DID Pattern and Caller ID Pattern.



- **Trunk**: Select the trunk that users will call in.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

  - **Default Destination**: Select the destination to the queue of Team-A.
  - **Time Condition**: Unselected.

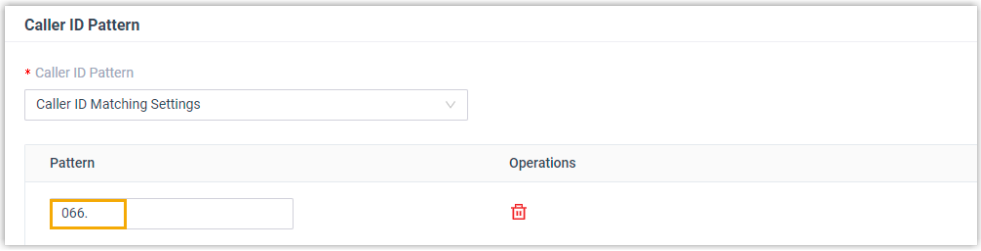


- **Fax Detection**: Leave the settings as default.

Inbound Route for Team-B

- **Name**: Enter a name to help you identify it.
- **DID Pattern**: Leave it blank, which means no limit of DID numbers.
- **Caller ID Pattern**: Select **Caller ID Matching Settings**, click **Add** and enter a Caller ID pattern or a full Caller ID.

  In this scenario, enter `066.`, which matches all inbound Caller IDs that start with digit 066. For more information of Caller ID pattern, see DID Pattern and Caller ID Pattern.

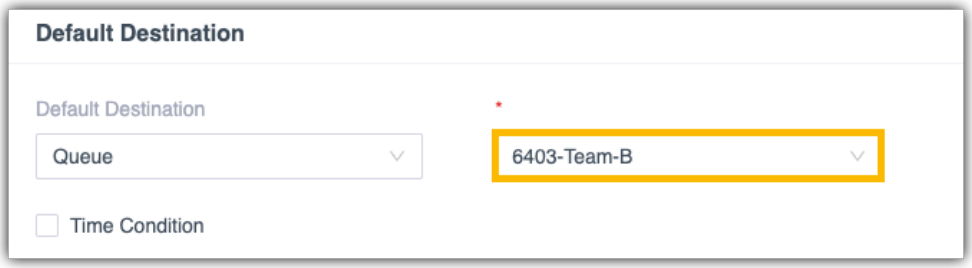


- **Trunk**: Select the trunk that users will call in.

  In this example, select **siptrunk**, whose phone number is 400-661-8815.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

  - **Default Destination**: Select the destination to the queue of Team-B.
  - **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

- When users from Fujian dial the number 400-661-8815, agents in Team-A will handle the calls.
- When users from Guangdong dial the number 400-661-8815, agents in Team-B will handle the calls.

Related information

Route Inbound Calls based on Global Business Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on Employee Hours
Route Inbound Calls based on DID Numbers
Route Inbound Calls by Matched Phonebook Contacts

Route Inbound Calls by Matched Phonebook Contacts

After grouping company contacts into phonebooks, you can set up inbound routes to distribute inbound calls from contacts to different destinations based on phonebooks.

Prerequisites

- You have added phonebooks and enabled **Caller ID Match** feature.

  For more information, see Manage Company Phonebooks and Identify Callers from Contacts.

Scenario

Company ABC has a Sales Team and a Support Team, both teams have their own customer groups. System administrator has added the customer information into two phonebooks.

Table 23.

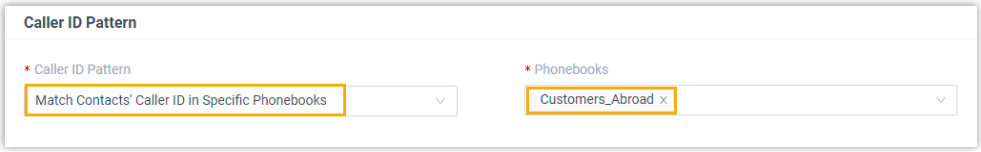
Team	Phonebook
Sales Team (Queue 6401)	Customers_Abroad
Support Team (Queue 6402)	Customers_China

Configuration Example

To distribute inbound calls from customers to corresponding team, you can set up two inbound routes to route calls by matching contacts in different phonebooks.

Inbound Route for Sales Team

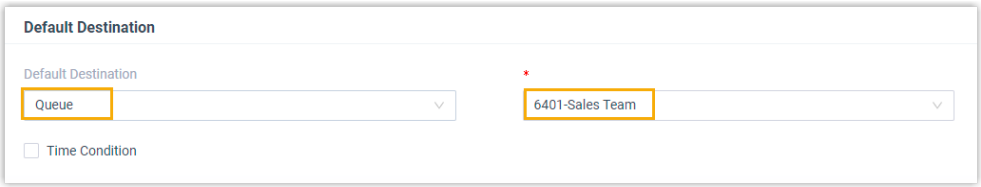
- **Name**: Enter a name to help you identify it.
- **DID Pattern**: Leave it blank, which means no limit of DID numbers.
- **Caller ID Pattern**: Select **Match Contacts' Caller ID in Specific Phonebooks** and select the phonebook **Customers_Abroad**.



- **Trunk**: Select the trunk that contacts will call in.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

  - **Default Destination**: Select the destination to **Queue** and select the Sales Team.
  - **Time Condition**: Unselected.

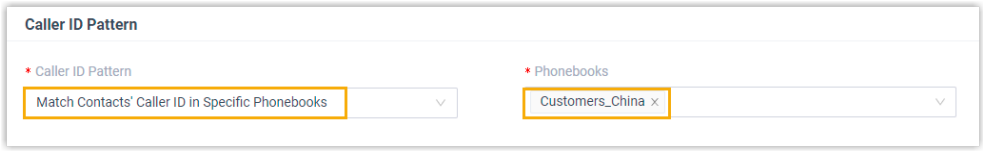


- **Fax Detection**: Leave the settings as default.

Inbound Route for Support Team

- **Name**: Enter a name to help you identify it.

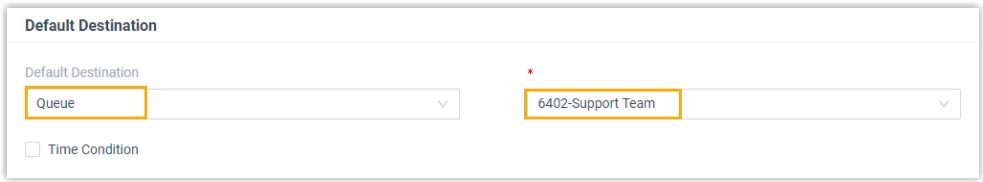
- **DID Pattern**: Leave it blank, which means no limit of DID numbers.
- **Caller ID Pattern**: Select **Match Contacts' Caller ID in Specific Phonebooks** and select the phonebook **Customers_China**.



- **Trunk**: Select the trunk that contacts will call in.
- **Default Destination**: Decide whether to route inbound calls to different destinations based on time and configure the destinations.

  In this example, route inbound calls to the default destination whenever the calls reach the system.

  - **Default Destination**: Select the destination to **Queue** and select Support Team.
  - **Time Condition**: Unselected.



- **Fax Detection**: Leave the settings as default.

Result

- When customers from Phonebook 'Customers_Abroad' call to PBX, Sales Team will handle the calls.
- When customers from Phonebook 'Customers_China' call to PBX, Support Team will handle the calls.

Related information

Route Inbound Calls based on Caller ID
Route Inbound Calls based on DID Numbers
Route Inbound Calls based on Employee Hours
Route Inbound Calls based on Department Hours
Route Inbound Calls based on Global Business Hours

Manage Inbound Routes

After you create inbound routes, you can adjust the priority of the inbound routes. You can also edit or delete the inbound routes.

Adjust priority of inbound routes

A trunk can be selected to multiple inbound routes. When users call to a trunk that is selected in multiple inbound routes, the system will route inbound calls through the route with higher priority. You can adjust the priority of inbound routes according to your needs.

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. In the Inbound Route list, click  to adjust the priority of your inbound routes.

Edit an inbound route

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. Click  beside the inbound route that you want to edit.
3. Edit the inbound route.
4. Click **Save** and **Apply**.

Delete an inbound route

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. Click  beside the inbound route that you want to delete.
3. On the pop-up window, click **Yes** to confirm.
4. Click **Apply**.

Export and Import Inbound Routes

The inbound routes configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired inbound routes in the exported file, and import the file to PBX again. This topic describes how to export and import inbound routes.

Export inbound routes

You can export all inbound routes to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Inbound Route Parameters.

Import inbound routes

We recommend that you export inbound route data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Inbound Route Parameters.

Procedures

1. Log in to PBX web portal, go to **Call Control > Inbound Route**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The inbound routes in the CSV file will be displayed in the **Inbound Route** list.

Related information

Import and Export -FAQ

DID Pattern and Caller ID Pattern

This topic describes special characters that can be defined in a DID pattern or a Caller ID pattern, and provides examples to help you understand and configure the pattern.

Pattern

A **Pattern** field appears when you are configuring DID numbers or Caller IDs. The **Pattern** field allows you to enter a full number or special characters that will match specific numbers.

The following table shows descriptions of the allowed characters in the Pattern field.

Table 24.

Pattern	Description
`X`	Match any digit from 0 -9.
`Z`	Match any digit from 1- 9.
`N`	Match any digit from 2 - 9.
`[###]`	Match any digit in the bracket. **Example**: `[123]` matches the numbers 1, 2, or 3. **Note:** Range of numbers can be specified with a dash, example `[136-8]` matches the numbers 1, 3, 6, 7, or 8.
`.`	Match one or more numbers. **Example**: `9011.` matches any numbers starting with digits 9011 (excluding 9011 itself).
`!`	Match none or more than one characters. **Example**: `9011!` matches any numbers starting with 9011 (including 9011 itself).

Pattern examples

The following table gives several patterns and list examples of matched numbers and mismatched numbers.

Table 25.

Pattern	Matched Number	Mismatched Number
0591.	• 05910 • 0591012345	• 0591 • 0592229929
+[13-5]XZN!	• +4022 • +1136282882	• +0136282882 • +1106282882
0591ZXXXX	• 059123456 • 059133456	• 05912345 • 059103456

Outbound Route

Outbound Route Overview

An outbound route tells the Yeastar P-Series Cloud Edition how to handle outbound calls based on pre-configured rules and criteria. When a user makes an outbound call, the system analyses the user's extension number and the dialed number, then routes the call through a matched outbound route.

Outbound Route matching criteria

Yeastar P-Series Cloud Edition provides the following criteria for you to configure outbound routes.

Dial Pattern

A dial pattern matches the dialed number and reformats the dialed number before sending the number out to the carrier.

For more information of dial patterns, see Outbound Dial Pattern.

Outbound Route Password

Users need to enter the PIN number before they can make calls through the outbound route.

Time Condition

A Time Condition defines when the outbound route is available.

Outbound Route priority

When a user makes an outbound call, the system compares the dialed number with the dial patterns in each outbound route (from highest to lowest priority) until a match is found.

- If the first outbound route is matched, the system will place the call through the outbound route.
- If the first outbound route is not matched, the system will check the second outbound route, and so on.

For more information, see Adjust priority of outbound routes.

Set up an Outbound Route

To allow users to make outbound calls through trunks, you need to set up at least one outbound route on the Yeastar P-Series Cloud Edition.

Background information

Yeastar P-Series Cloud Edition has a default outbound route with dial pattern `X.` that allows users to dial any outgoing numbers. You can delete the default outbound route, then add a new one to configure settings according to your needs.

Prerequisites

Ensure that you have set up at least one trunk for outbound calls.

Procedure

1. Log in to PBX web portal, go to **Call Control > Outbound Route**, click **Add**.
2. In the **General** section, complete the following configurations:
   - **Name**: Enter a name to help you identify it.
   - **Outbound Caller ID**: Optional. By default, each trunk is associated with a main caller ID. When users make outbound calls through a trunk, the main caller ID is displayed on the called party's device. If this option is configured, the system will override the main caller ID with the Outbound Caller ID.

     For more information of caller ID, see Caller ID Overview.

     > **Note:**
     > Only configure this setting when the trunk provider supports Caller ID override, or the following errors may happen:
     > - Outbound calls failed to be established.
     > - Caller ID doesn't be overridden.
3. In the **Dial Pattern** section, configure dial rules for the outbound route.
   a. Click **Add**.
   b. Configure the dial pattern to match dialed numbers and reformat dialed numbers.
      - **Pattern**: Enter a pattern to match dialed numbers. Only when the dialed number is matched will the call go through this outbound route.
      - **Strip**: Optional. To strip digits from the beginning of the dialed numbers, enter a value in this field to define how many digits will be removed.

- **Prepend**: Optional. To add digits at the beginning of the dialed number, enter the digits that you want to prepend in this field.

c. To add more dial patterns, repeat **step a-b**.

4. In the **Trunk** section, configure the followings:
   a. Select one or more trunks from the **Available** box to **Selected** box.
   b. **Optional:** If multiple trunks are selected, configure the trunk sequence.
      - **Default trunk sequence**

        Click the buttons ⤒ ^ v ⤓ beside the **Selected** box to specify the default trunk sequence. By default, the system always selects an idle trunk from top to bottom, and uses the trunk to call out.
      - **Rrmemory Hunt**

        If the option **Rrmemory Hunt** is selected, the system will remember which trunk was used last time, and use the next idle trunk to call out.

   c. To enhance the outbound route security, configure the **Outbound Route Password**.
      - **Disable**: No password is required to call out through this outbound route.
      - **Single PIN**: Set a single PIN. All the users need to enter the same PIN to make outbound calls through this outbound route.
      - **PIN List**: Select a PIN list. Users are required to dial a password included in this list before an outbound call go through.

        > **Note:**
        > Generally, each user has a specific PIN code assigned by the administrator. For more information, see Add a PIN List.

5. Select which users are allowed to make calls through this outbound route.
   In the **Extension/Extension Group** section, select extensions, extension groups, or organizations from **Available** box to **Selected** box.

   > **Note:**
   > - Organizations are displayed only when you enable the **Organization Management** feature.
   > - By default, when you select an organization, its associated sub-organizations are selected. Be careful when selecting organizations.

6. **Optional:** In the **Time Condition** section, select an option from **Available Time** drop-down list to specify when this outbound route is available to use.

- **Always**: This outbound route is available at any time for allowed extension users.
- **Based on Global Business Hours**: Set up whether to allow this route in the following time separately:
  - **Business Hours**: Global Business Hours specified in the system.
  - **Holidays**: Holidays specified in the system.
  - **Outside Business Hours**: The time periods that are not defined as Business Hours or Holidays.
- **Based on Custom Business Hours**: Set up custom business hours and configure whether to allow this route in the following time:
  - **Business Hours**: The custom business hours.
  - **Holidays**: Holidays specified in the system.
  - **Outside Business Hours**: The time periods that are not defined as Business Hours or Holidays.
- **Based on Custom Time Periods**: Set up multiple time periods for this route. You can also specify whether to allow this route in the Holidays.

7. Click **Save** and **Apply**.

What to do next

After you finish the outbound route configurations, you need to check and adjust the priority of your outbound routes, so that the system can match and route the call out through the proper outbound route.

For more information, see Adjust priority of outbound routes.

Restrict Outbound Calls by PIN Codes

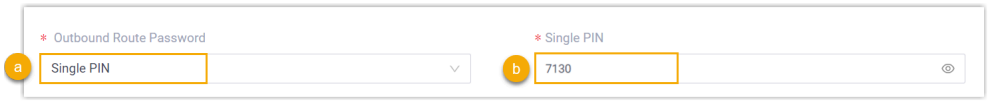
To prevent the abuse of communication resources, you can restrict outbound calls by setting password for outbound routes to require callers to enter a PIN code before dialing out. Only when a valid PIN code is entered can the call be sent out through the outbound route. This topic describes how to configure either a single PIN code or a list of PIN codes for an outbound route.

Restrict outbound calls by a single PIN code

To allow only specific employees or departments to make outbound calls via a specific outbound route, you can set a unique PIN code for the route. In this way, only users with the correct PIN code are able to use the outbound route.

Procedure

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click beside the desired outbound route.
3. Scroll down to the **Trunk** section, set up a password for the outbound route.



   a. In the **Outbound Route Password** drop-down list, select **Single PIN**.
   b. In the **Single PIN** field, set a PIN code.

> **Note:**
> - Only allows numeric value.
> - The length of the PIN code is limited from 3 to 15.

4. Click **Save** and **Apply**.

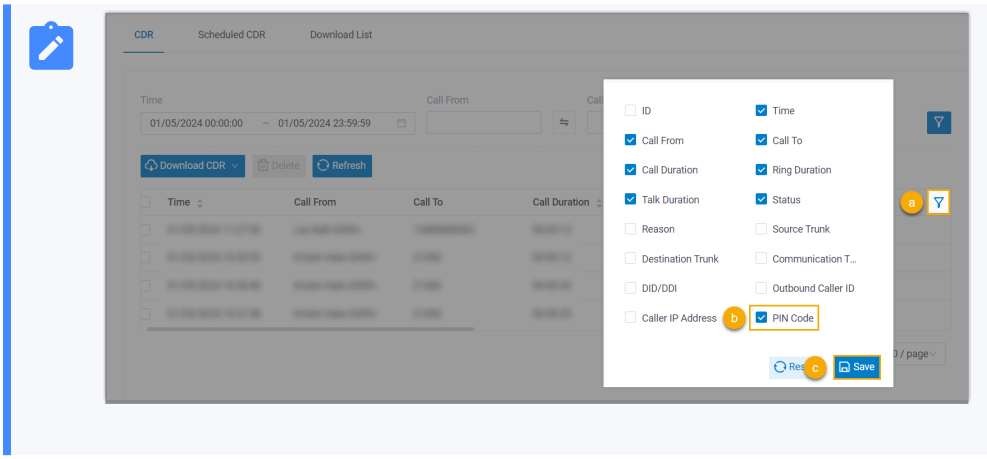
Result

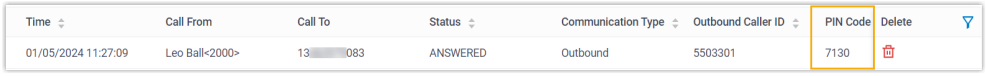
- Users need to enter the PIN code when making outbound calls via the outbound route.

> **Note:**
> If a user enters a wrong PIN code for three times, the call will be hung up automatically.

- In the CDR, you can see the PIN code used in the outbound calls made via the outbound route (Path: **Reports and Recordings > CDR**).

> **Note:**
> Make sure that you have set the CDR to display the **PIN Code** column.





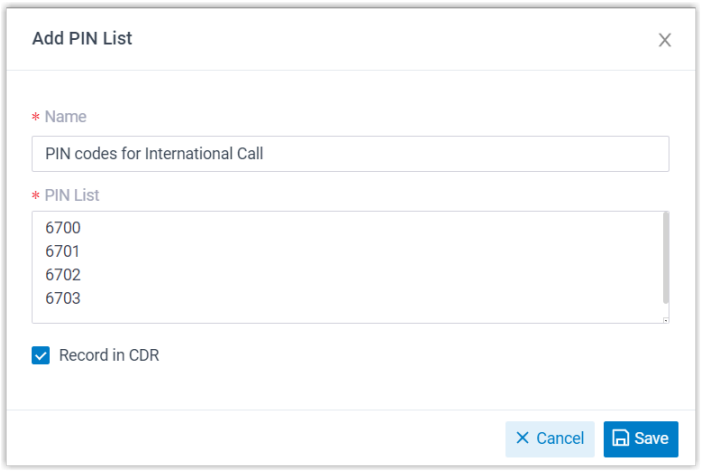
For more information about how extension users can use the PIN code for making outbound calls, see the usage example.

Restrict outbound calls by multiple PIN codes

In the scenario where a number of users share the same extension number across different endpoints to make outbound calls, it is difficult to track the calling activities of these users individually. In this case, you can set up a list of passwords for an outbound route and allocate them to users, so that users can make outbound calls via the outbound route, while you can track and distinguish their calls using the PIN codes.

Step 1. Add a PIN List

1. Log in to PBX web portal, go to **Call Features > PIN List**, click **Add**.
2. In the pop-up window, configure the following settings:

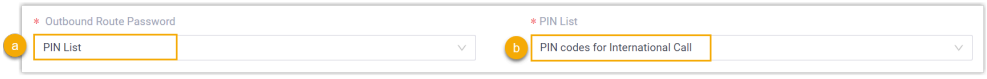


Setting	Description
Name	Enter a name to help you identify it.
PIN List	Enter PIN codes. **Note:** • Press the **Enter** key to separate multiple PIN codes. • Only allows numeric value. • The length of each PIN code is limited from 3 to 15.
Record in CDR	Enable this option. The system will record the PIN code in CDR when the PIN code has been used.

3. Click **Save**.

Step 2. Associate the PIN List with an outbound route

1. On PBX web portal, go to **Call Control > Outbound Route**.
2. Click beside the desired outbound route.
3. Scroll down to the **Trunk** section, set up passwords for the outbound route.



   a. In the **Outbound Route Password** drop-down list, select **PIN List**.

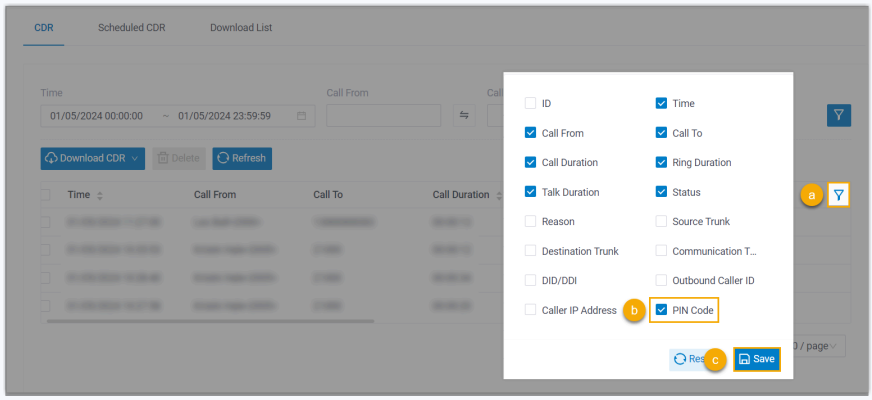
b. In the **PIN List** drop-down list, select an existing PIN list.

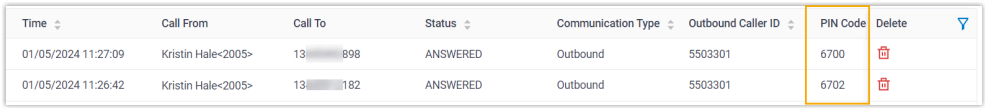
4. Click **Save** and **Apply**.

Result

- Users need to enter any PIN codes contained in the selected PIN list when making outbound calls via the outbound route.
- In the CDR, you can easily distinguish the users who made outbound calls via different PIN codes allocated to different users, even if they use the same extension to dial out (Path: **Reports and Recordings > CDR**).

Note:
Make sure that you have set the CDR to display the **PIN Code** column.





Time	Call From	Call To	Status	Communication Type	Outbound Caller ID	PIN Code	Delete
01/05/2024 11:27:09	Kristin Hale<2005>	13[redacted]898	ANSWERED	Outbound	5503301	6700	
01/05/2024 11:26:42	Kristin Hale<2005>	13[redacted]182	ANSWERED	Outbound	5503301	6702	

For more information about how extension users can use the PIN code for making outbound calls, see the usage example.

Usage example

We provide an example to help you understand the workflow of making outbound calls via a restricted outbound route using a PIN code.

1. An extension user 2000 dials an external number.

2. After hearing a voice prompt "Please enter your password followed by the pound key", the user enters a PIN code. When done, press # key.
    - If the PIN code is correct, the call will be successfully sent out through the outbound route.
    - If the PIN code is incorrect, the user will hear a voice prompt "password incorrect" and will be asked to enter a password again. If the user enters wrong password for three times, the call will be hung up automatically.

Manage Outbound Routes

After you create outbound routes, you can adjust the priority of the outbound routes. You can also edit or delete the outbound routes.

Adjust priority of outbound routes

When a user places a call, if the dialed number matches multiple dial patterns, the outbound route with the highest priority will be used. You can adjust the priority of outbound routes to route calls through proper outbound routes.

> **Note:**
> The route priority is important, especially if there is some overlap. For example, the number 5503305 matches both dial patterns of `ZXXXXXX` and `X.`, the PBX will send the call through the outbound route with the highest priority.

Example:

When users dial 05503301, both of the two outbound routes match 05503301:

- Outbound Route-Long-distance call: The dial pattern is `0XXXXXXX` and uses trunk 1.
- Outbound Route-Local call: The dial pattern is `X.` and uses trunk 2.

To call 5503301 through trunk 1, you need to prioritize the outbound route of "Long-distance call"; or PBX will match the outbound route of “Local call" and route the call out using trunk 2.

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click the buttons ⤒ ^ v ⤓ to adjust the priority of your outbound routes.

> **Note:**

PBX will match outbound route from top to bottom.

Edit an outbound route

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click beside the inbound route that you want to edit.
3. On the outbound route configuration page, edit the outbound route.
4. Click **Save** and **Apply**.

Delete an outbound route

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click beside the outbound route that you want to delete.
3. On the pop-up dialog box, click **OK** to confirm.
4. Click **Apply**.

Export and Import Outbound Routes

The outbound routes configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired outbound routes in the exported file, and import the file to PBX again. This topic describes how to export and import outbound routes.

Export outbound routes

You can export all outbound routes to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Outbound Route Parameters.

Import outbound routes

We recommend that you export outbound route data to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Outbound Route Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Control > Outbound Route**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The outbound routes in the CSV file will be displayed in the **Outbound Route** list.

Related information

Import and Export -FAQ

Outbound Dial Pattern

This topic describes dial pattern settings of Outbound Route.

Dial Pattern components

A dial pattern comprises **Pattern**, **Strip**, and **Prepend**.

Pattern

Required.

Defines which dialed numbers will be matched.

The Pattern field allows a full number or special characters that will match specific numbers. The following table shows descriptions of the allowed characters in the Pattern field.

Pattern	Description
X	Match any digit from 0 -9.
Z	Match any digit from 1- 9.

Pattern	Description
`N`	Match any digit from 2 - 9.
`[###]`	Match any digit in the bracket. **Example**: `[123]` matches the numbers 1, 2, or 3. **Note:** Range of numbers can be specified with a dash, example `[136-8]` matches the numbers 1, 3, 6, 7, or 8.
`.`	Match one or more numbers. **Example**: `9011.` matches any numbers starting with digits 9011 (excluding 9011 itself).
`!`	Match one or more characters. **Example**: *9011!* matches any numbers starting with 9011 (including 9011 itself).

Strip

Optional.

Defines how many digits will be stripped from the beginning of a dialed number when the dialed number successfully matches a **Pattern**.

Example:

If you set **Pattern** as *9.* and set **Strip** as *1*.

If a user wants to call number 1588902923, the user should dial 91588902923. The PBX will strip digit 9 from the dialed number, and call the number 1588902923.

Note:

- The system strips leading digits before sending the number to the carrier.
- If both **Strip** and **Prepend** are configured, the system first strips leading digits from the dialed number then prepends digits to the dialed number.

Prepend

Optional.

Defines which digits will be added at the beginning of a dialed number when the dialed number successfully matches a **Pattern**.

Example:

202 is the area code for Washington, D.C. For users who often make calls to the city, you can set **Prepend** as **202**.

In this case, if a user wants to call number 2025553097, the user should dial 5553097.

Note:

- The system prepends the digits before sending the number to the carrier.
- If both **Strip** and **Prepend** are configured, the system first strips leading digits from the dialed number then prepends digits to the dialed number.

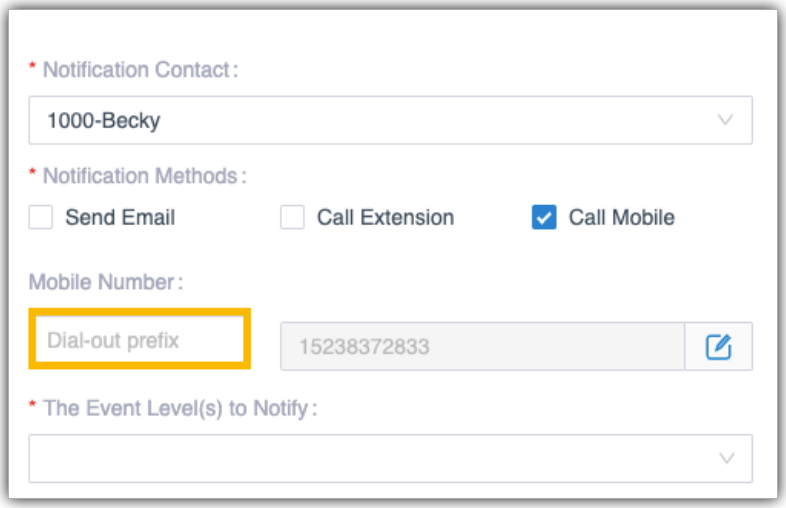
Prefix and Dial Pattern

A prefix is the digit that will be removed from the dialed number before sending to the carrier.

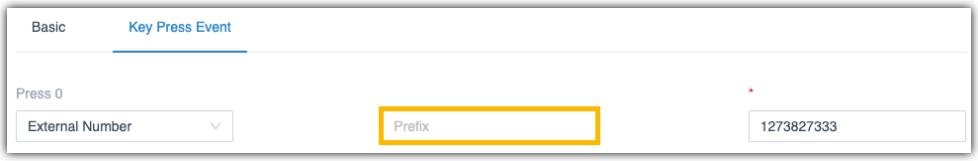
Scenarios

Prefix setting appears when you are configuring the following settings:

- Mobile phone number for notification contacts.



- External number for IVR keypress.



How to configure Prefix

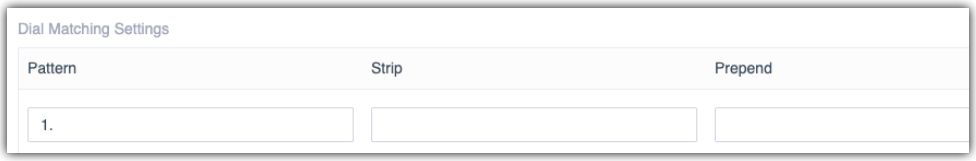
You need to configure prefix according to the dial pattern settings on your out-bound route. If the prefix is not configured correctly, the PBX cannot call to the external number successfully.

- **Leave Prefix setting blank**

  If the **Strip** of outbound route is not set, you don't have to add a prefix before the phone number.

  As the following figure shows, only the destination number that starts with digit *1* can be called out through this outbound route.

  For example, to call number 125451, you should dial the number 125451 directly.

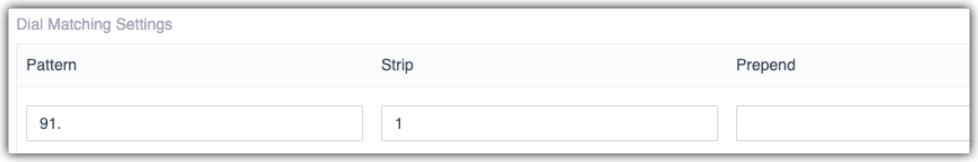


- **Add prefix before a number**

If **Strip** is set on an outbound route, you need to set the prefix according to the **Pattern**.

As the following figure shows, to make calls through the outbound route, you need to add prefix *9* before the number, and the destination number should start with digit *1*.

For example, to call number 125451, you should add prefix 9 before the number 125451.



Dial Pattern Examples

This topic provides sample dial patterns to help you understand dial patterns of outbound route.

Local calls

In Xiamen, China, local numbers are all 7-digit numbers and the numbers do not start with 0, such as 5503305.

For the local calls, set dial pattern as the following table shows.

Pattern	Strip	Prepend
ZXXXXXX	Leave it blank.	Leave it blank.

Long distance calls

In Xiamen, China, users need to dial 4-digit area code and 7-digit local number to make a long distance call, such as 0595-7588123.

- Area code format: 0ZXX, the first digit is 0, and the second digit cannot be 0.
- Local number format: 7-digit number that does not start with 0.

For long distance calls, set dial pattern as the following table shows.

Pattern	Strip	Prepend
0ZXXZXXXXXX	Leave it blank.	Leave it blank.

Mobile calls

All mobile phone numbers in China are 11-digit numbers and start with digit 1, such as 15880260666.

For mobile calls, set dial pattern as the following table shows.

Pattern	Strip	Prepend
1XXXXXXXXXX	Leave it blank.	Leave it blank.

International calls

All international numbers start with digits 00.

For international calls, set dial pattern as the following table shows.

Pattern	Strip	Prepend
00.	Leave it blank.	Leave it blank.

AutoCLIP Route

AutoCLIP Route Overview

Yeastar provides AutoCLIP (Auto Calling Line Identification Presentation) feature, which is an intelligent call matching feature. You can configure AutoCLIP to route customer inbound calls to original extensions, which will promote your customer satisfaction and help your business be more efficient, and professional.

Restriction

The AutoCLIP list supports up to 100,000 records.

Scenarios

Assume that sales representatives in your company often make outbound calls to customers for promotion. More or less, some customers may miss the calls. When customers call back, the calls are routed to the receptionist or business auto attendant. Neither receptionist/business auto attendant nor the customers know who placed the call.

With AutoCLIP feature, the PBX can redirect the calls to the original extension users who placed the calls when customers call back. Using this feature, you can avoid the embarrassing situation that customers cannot find the person when they call back to the PBX.

How does the PBX redirect calls to original extensions?

1. When extension users make outbound calls, the PBX automatically stores the records to AutoCLIP list, including extension number, called number, and the used trunk.
2. When customers call back to the PBX system, PBX will compare the phone numbers with the records in the AutoCLIP list.
    - If there're matched records in AutoCLIP list, the calls will be routed to corresponding extensions, bypassing any receptionists or business auto attendant.
    - If there're not matched records in AutoCLIP list, the calls will be routed to the destination specified in inbound routes.

Route Inbound Calls to Original Extensions via AutoCLIP Route

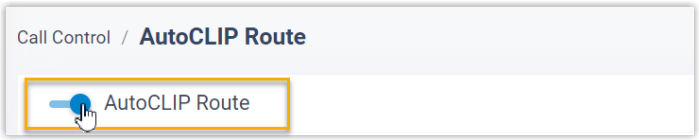
With AutoCLIP feature, Yeastar P-Series Cloud Edition can route inbound calls from customers to original extension users who placed the calls. This intelligent call matching feature can greatly improve work efficiency and customer satisfaction. This topic describes how to set up the AutoCLIP route.

Prerequisites

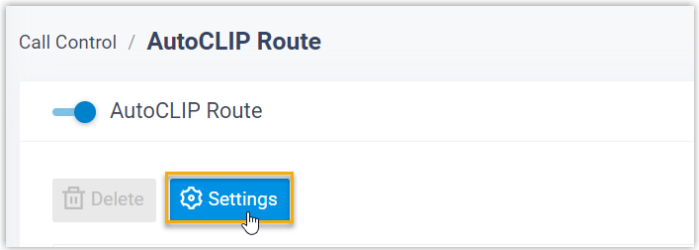
Make sure the desired trunk purchased from trunk provider has the Caller ID feature, or the PBX can not distinguish the Caller ID and implement AutoCLIP.

Procedure

1. Log in to PBX web portal, go to **Call Control > AutoCLIP Route**.
2. On the top of the page, enable the **AutoCLIP Route** feature.



3. Click **Settings** to set up rules for AutoCLIP route.



4. Configure the AutoCLIP settings according to your needs.

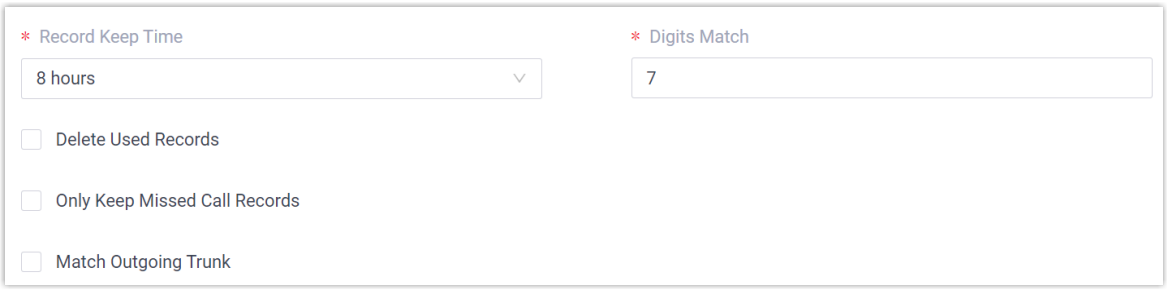
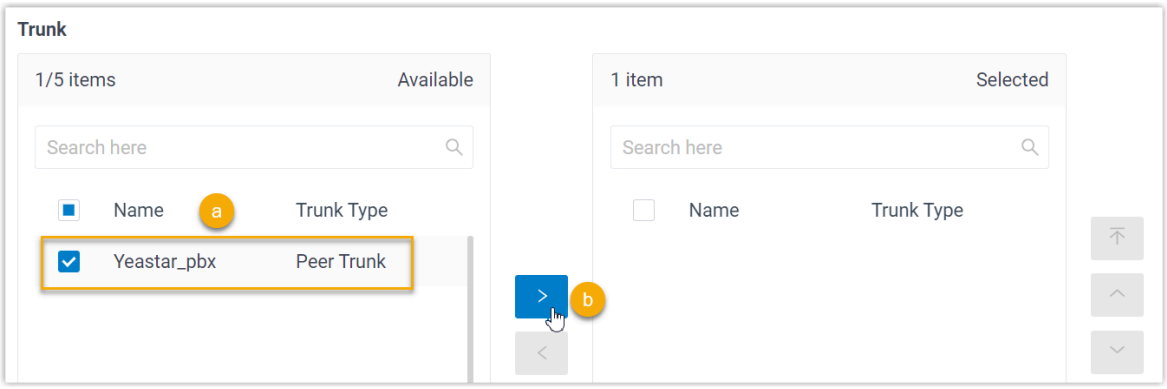


Table 26.

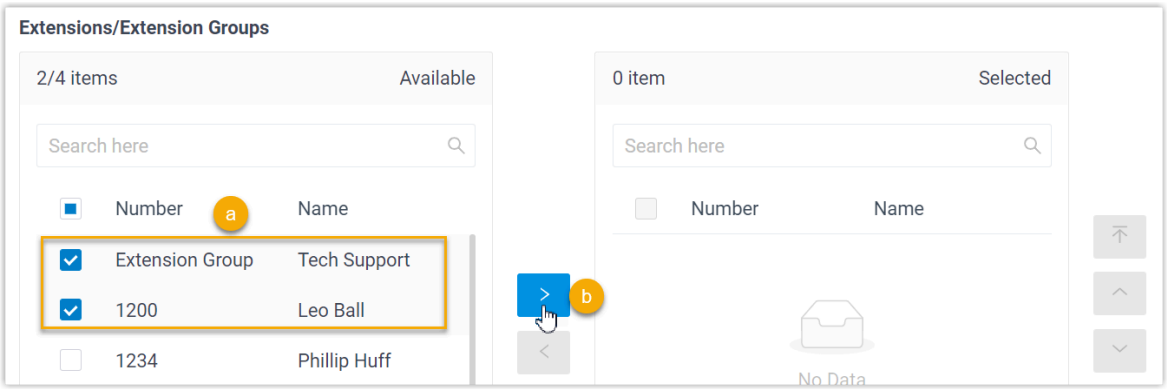
Setting	Description
Record Keep Time	Set how long records can be kept in AutoCLIP list. If keep time of a record exceeds the value, PBX will automatically delete the record. **Tip:** You can check the expiration time in the AutoCLIP record list directly.
Digits Match	Define how many digits from the last digit of the incoming Call ID will be used to match the AutoCLIP list. **Note:** If the number has fewer digits than the value defined here, it will be matched in full length.
Delete Used Records	If enabled, when an AutoCLIP record is matched, it will be deleted from the record list automatically after the original extension has answered the redirected customer call.
Only Keep Missed Call Records	If enabled, only when the outbound call is not answered will it be recorded in the AutoCLIP list.
Match Outgoing Trunk	If enabled, the PBX will route the call to the original extension only when the trunk number dialed by external users matches the trunk that was used to place the call earlier.

5. In the **Trunk** section, select which trunks will use AutoCLIP Route.



a. Select the desired trunk(s).
b. Add the desired trunk(s) from **Available** box to **Selected** box.

6. In the **Extensions/Extension Groups** section, select which extensions can use Auto-CLIP Route.



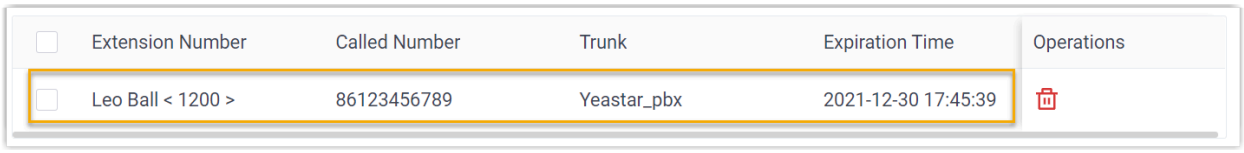
a. Select the desired extension(s)/extension group(s).
b. Add the extension(s)/extension group(s) from **Available** box to **Selected** box.

7. Click **Save**.

Result

When extension user uses the trunk with AutoCLIP feature to call external users out, PBX generates AutoCLIP records, including extension details, the numbers dialed and the used trunk. You can check the AutoCLIP record on **Call Control > AutoCLIP Route**.

Note:
If more than one extension user make outbound calls to the same external user, PBX will only match the last extension user that placed the call when the external user calls back.



Delete AutoCLIP Records

This topic describes how to delete AutoCLIP records.

Delete a record

1. Log in to PBX web portal, go to **Call Control > AutoCLIP Route**.
2. Click beside the record that you want to delete.
3. In the pop-up dialog box, click **OK**.

Bulk delete records

1. Log in to PBX web portal, go to **Call Control > AutoCLIP Route**.
2. Select the checkboxes of the desired records, and click **Delete**.
3. In the pop-up dialog box, click **OK**.

DID Number

DID Number Overview

This topic describes what is DID number and DID usages on Yeastar P-Series Cloud Edition.

What is a DID number?

Direct Inward Dialling (DID), also called Direct Dial-in (DDI), is a service offered by telephone companies. A telephone company usually assigns a range of numbers to a trunk. There is an extra charge for the DID numbers, you need to contact the trunk provider to purchase DID numbers.

DID usages

Yeastar P-Series Cloud Edition allows you to configure DID numbers on an inbound route or a trunk to achieve different functions.

DID configuration on an inbound route

- A company can use DID numbers to identify incoming calls of different purposes, such as incoming calls for customer service, sales, etc.
- DID numbers can also be assigned to individual employees. In this way, callers can dial directly into extension users on the Yeastar P-Series Cloud Edition.

For more information, see Route Inbound Calls based on DID Numbers.

DID configuration on a trunk

- **For SIP Register Trunk**

  For a SIP Register Trunk, if ITSP provides DID numbers that are different from SIP authentication name, you need to add the provided DID numbers on the trunk, or inbound calls through this trunk would fail.
- **Identify inbound calls**

  To identify which DID number is dialed, you can bind each DID number with a DID name.

For more information, see Configure DID Numbers on a Trunk.

Configure DID Numbers on a Trunk

This topic describes when and how to configure DID numbers on a trunk.

Background information

DID numbers are usually configured on inbound routes to distinguish inbound calls. For more information, see Route Inbound Calls based on DID Numbers.

In the following scenarios, you need to configure DID numbers on a trunk:

- **For SIP Register Trunk**

  For a SIP Register Trunk, if ITSP provides DID numbers that are different from SIP authentication name, you need to add the provided DID numbers on the trunk, or inbound calls through this trunk would fail.
- **Identify inbound calls**

  To identify which DID number is dialed, you can bind each DID number with a DID name.

Prerequisites

Purchase DID numbers from the trunk provider.

Add a DID number

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Click **DIDs/DDIs** tab.
3. In the pop-up window, click **Add** and configure the DID.
   - **DID/DDI**: Enter the provided DID number.
   - **DID/DDI Name**: Bind a name with the DID number.

     When the DID number is dialed, the name will be displayed on the called party's device.
4. Click **Save** and **Apply**.

Delete DID numbers

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Click **DIDs/DDIs** tab.
3. On the **DIDs/DDIs** page, click  to delete a DID number.
4. To bulk delete DID numbers, select the checkboxes of DID numbers, click **Delete**.
5. Click **Save**.

Export and Import Trunk DIDs/DDIs Numbers

Trunk DIDs/DDIs numbers configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired DIDs/DDIs numbers in the exported file, and import the file to PBX again. This topic describes how to export and import DIDs/DDIs numbers.

Export all DIDs/DDIs numbers

You can export all DIDs/DDIs numbers to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. In the **DIDs/DDIs** tab, click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Trunk DIDs/DDIs Parameters.

Import DIDs/DDIs numbers

We recommend that you export DIDs/DDIs numbers to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 5 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Trunk DIDs/DDIs Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. In the **DIDs/DDIs** tab, click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The DIDs/DDIs numbers in the CSV file will be displayed in the **DIDs/DDIs** list.

Related information

Import and Export -FAQ

Caller ID

Caller ID Overview

This topic describes what is caller ID, differences between all the types of caller ID defined in Yeastar P-Series Cloud Edition.

What is Caller ID?

Caller ID is a telephone service that transmits a caller's telephone number and name to the called party's device when a call is established.

Caller ID types

Yeastar P-Series Cloud Edition supports the following types of Caller ID:

Outbound caller ID

Outbound caller ID is the phone number that will be displayed on the called party's phone when an extension user makes an outbound call. Each trunk has a main number, the number appears when an outbound call is received by a recipient.

To customize the outbound caller ID, you need to purchase the service from the trunk provider, and set the custom outbound caller ID on the PBX. In Yeastar P-Series Cloud Edition, you can configure outbound caller ID based on the following features:

- Emergency Numbers
- Outbound Route
- Trunk
- Extension

For more information, see Customize Outbound Caller IDs.

Inbound caller ID

Inbound caller ID is an external user's phone number that will be displayed on an extension user's phone when the external user calls in Yeastar P-Series Cloud Edition.

Inbound Caller IDs can be reformatted before they are sent to the destination users. For more information, see Reformat Inbound Caller ID based on a Trunk.

Priority of outbound caller ID

When an extension user makes an outbound call, the system first identifies if the call is an emergency call, then sends an outbound caller ID by the following priority (from the highest to the lowest).

1. Extensions' emergency outbound caller ID
2. Trunk's emergency outbound caller ID
3. Outbound Route caller ID
4. Trunk's outbound caller IDs that are associated with extension users

5. Trunk's general outbound caller ID
6. Trunk's default phone number that is provided by the carrier
7. Extension's caller ID

Reformat Inbound Caller ID based on a Trunk

This topic describes how to reformat inbound caller ID and gives configuration examples to help you understand the reformatting rule.

Background information

If an inbound caller ID is in the format that is inconvenient for users to redial directly, you can reformat the inbound caller ID.

Reformatting inbound caller ID is supported on all types of trunk. Based on different trunk providers, you may need to set up different rules to reformat inbound caller IDs.

Add a rule to reformat inbound Caller ID

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Click **Inbound Caller ID Reformatting** tab.
3. On the **Inbound Caller ID Reformatting** page, click **Add**.
4. In the pop-up window, configure the reformatting rule and click **Confirm**.
   - **Patterns**: Specify which Caller IDs will be reformatted. The inbound caller ID that matches this pattern will be reformatted.
   - **Strip**: Specify how many digits will be stripped from the beginning of the inbound caller ID.
   - **Prepend**: Specify the digits that will be prepended to the inbound caller ID.

   > **Note:**
   > If both **Strip** and **Prepend** are configured, the system will first strip the leading digits then add the prepend digits to the inbound caller ID.
5. Click **Save** and **Apply**.

Example 1

Company A wants to add a digit 0 to the 11-digit inbound caller ID number that begins with digit 1 for quick redial purpose.

For example, company A wants to display 012345678910 instead of 12345678910.

In this case, you can configure the reformatting rule as below:



- **Patterns**: 1XXXXXXXXXX
- **Strip**: Leave it blank.
- **Prepend**: 0

Example 2

Company B wants all local numbers to be displayed without area code (0592).

For example, company B wants to display number 5503301 instead of 05925503301.

In this case, you can configure the reformatting rule as below:



- **Patterns**: 0592XXXXXXX
- **Strip**: 4
- **Prepend**: Leave it blank.

Export and Import Inbound Caller ID Reformatting Rules

The inbound caller ID reformatting rules configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired inbound caller ID reformatting rules in the exported file, and import the file to PBX again. This topic describes how to export and import inbound caller ID reformatting rules.

Export all inbound caller ID reformatting rules

You can export all inbound caller ID reformatting rules to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. In the **Inbound Caller ID Reformatting** tab, click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Inbound Caller ID Reformatting Rule Parameters.

Import inbound caller ID reformatting rules

We recommend that you export inbound caller ID reformatting rules to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 5 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Inbound Caller ID Reformatting Rule Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. In the **Inbound Caller ID Reformatting** tab, click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

The inbound caller ID reformatting rules in the CSV file will be displayed in the **Inbound Caller ID Reformatting** list.

Related information

Import and Export -FAQ

Customize Outbound Caller IDs

This topic describes different ways to customize outbound caller IDs, which help customers recognize who's calling.

Background information

Before you start to customize outbound caller IDs, you may need to know the following concepts:

- Caller ID types
- Priority of outbound caller ID

Prerequisites

Customizing outbound caller ID should be supported by the trunk provider.

Customize outbound caller ID for a trunk

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Click **Outbound Caller ID** tab.
3. In the **General** section, configure a general **Outbound Caller ID** and **Outbound Caller ID Name** for the trunk.
4. Click **Save** and **Apply**.

   The general outbound caller ID and caller ID name will be displayed on the called party's device when users make outbound calls through this trunk.

Customize outbound caller IDs for extensions

You can set up an outbound caller ID for a specific extension based on a trunk, so that an associated caller ID is sent out when the user calls out.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Click **Outbound Caller ID** tab.

3. In the **Outbound Caller ID List** section, click **Add**, and configure outbound caller IDs for extensions by different methods.
4. To associate one outbound caller ID with multiple extensions, select **Shared Outbound Caller ID** and configure the following settings:
    - **Outbound Caller ID**
    - **Outbound Caller ID Name**
    - **Associated Extensions**
5. To bind consecutive outbound caller IDs to consecutive extensions with one-to-one correspondence, select **Outbound Caller ID Range** and configure the following settings:
    - **Outbound Caller ID Range**
    - **Extension Range**
    - **Outbound Caller ID Name**
6. Click **Save** and **Apply**.

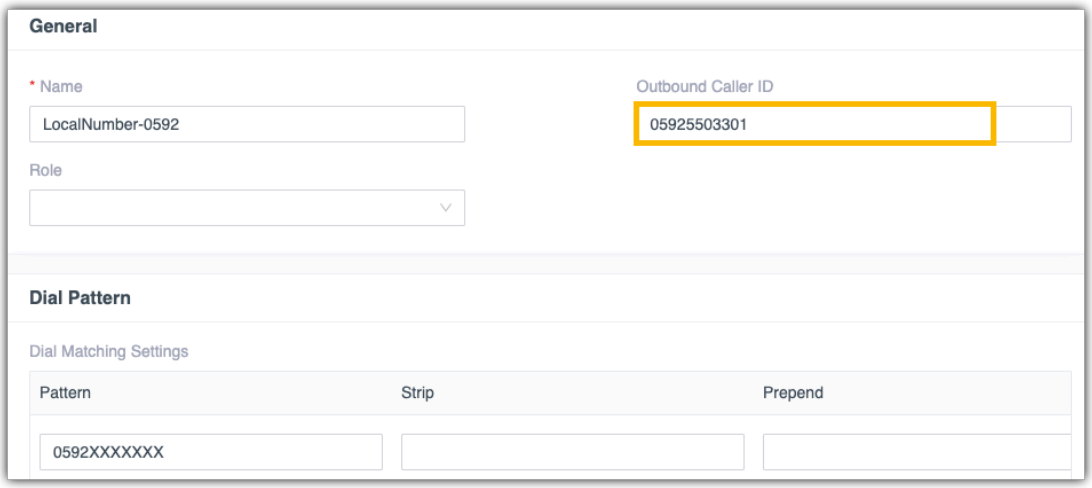
When extension users make outbound calls through the configured trunk, the associated outbound caller IDs will be displayed on the called party's device.

Customize outbound caller IDs based on dialed numbers

When calling to multiple areas, you may need to display pre-defined local number for the area code you are dialling. In this case, you can configure outbound caller IDs based on the dialed numbers.

The following instruction describes how to display a custom outbound caller ID 05925503301 when users call to local numbers that have area code 0592.

1. Log in to PBX web portal, go to **Call Control > Outbound Route**, edit the outbound route that is for local calls with area code 0592.
2. In the **General** section, enter the custom caller ID in the **Outbound Caller ID** field.



3. Click **Save** and **Apply**.

Export and Import Trunk Outbound Caller IDs

Trunk outbound caller IDs configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired outbound caller ID list numbers in the exported file, and import the file to PBX again. This topic describes how to export and import outbound caller ID list.

Export all outbound caller ID list

You can export all outbound caller ID list to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. Click the **Outbound Caller ID** tab.
3. In the **Outbound Caller ID List** section, click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Trunk Outbound Caller ID Parameters.

Import outbound caller ID list

We recommend that you export outbound caller ID list to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 5 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Trunk Outbound Caller ID Parameters.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit a desired trunk.
2. In the **Outbound Caller ID List** section, click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The outbound caller id in the CSV file will be displayed in the **Outbound Caller ID List**.

Related information

Import and Export -FAQ

Distinctive Ringtone

Distinctive Ringtone Overview

This topic describes what is Distinctive ringtone, applications, and how does Distinctive ringtone work.

What is Distinctive ringtone

Distinctive ringtone is an effective feature for businesses. Distinctive ringtone allows employees to distinguish incoming calls without looking at the Caller Name or Caller ID on the phone display.

For example, company may have different ring groups or queues for the sales team, the customer service team, and the support team. This could all be fed from IVR where the caller presses 1, 2, or 3 that equates to each team. For smaller businesses that have the same employee answering most of the calls, separating each business by its own distinctive ringtone can make the employee quickly identify who is calling or if the call is for him/her.

Important:
Distinctive ringtone is not support on all SIP phones. Make sure that your phones support playing distinctive ringtone by "alert info text".

Applications

With the Distinctive ringtone feature, you can assign different call ringtones for the following types of calls:

- Set Distinctive Ringtones for Internal Calls
- Set Distinctive Ringtones for External Calls
- Set Distinctive Ringtones for Queue Calls
- Set Distinctive Ringtones for Ring Group Calls
- Set Distinctive Ringtones for IVR Calls

How does Distinctive ringtone work

Distinctive ringtone feature allows certain incoming calls to trigger IP phones to play specific ringtones. The achievement of distinctive ringtone relies on an "alert info text".

1. Yeastar P-Series Cloud Edition adds an "alert info text" in Alert-Info header for incoming calls, and then sends the incoming call (an INVITE request with the Alert-Info header) to the IP phone.
2. The IP phone inspects the INVITE request for an "Alert-Info" header, strips out the "alert info text", and then plays corresponding ringtone associated with the "alert info text".

Set Distinctive Ringtones for Internal Calls

When an extension user hears the ringtone of an internal incoming call, the user may notice the intention of the call.

Procedure

1. Set alert info for internal calls on the PBX.
2. Set a specific ringtone for a phone.

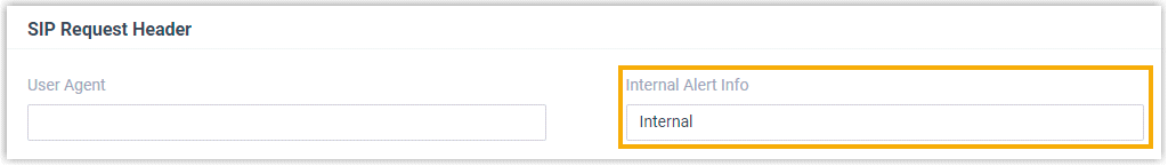
Set alert info for internal calls on the PBX

1. Log in to PBX web portal, go to **PBX Settings > SIP Settings > Advanced**.

2. In the **SIP Request Header** section, enter an alert info in the **Internal Alert Info** field.

   The alert info is used to trigger IP phones to play a specific ringtone when receiving an internal call.

   In this example, set alert info to `Internal`.



Set a specific ringtone for a phone

For users who want to play a specific ringtone for internal calls on their phones, you can set a specific ringtone for their phones by auto provisioning.

> **Note:**
> Users can also log in to phone web interface to set distinctive ringtone manually on their own IP phones. For more information, contact the phone manufacturer.

Prerequisites

Each user who wants distinctive ringtone has bound a phone with their extensions.

For more information, see Auto Provision IP Phones.

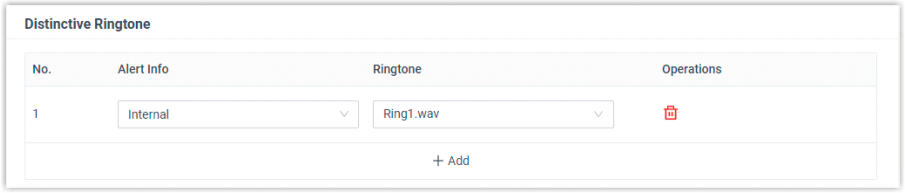
Procedure

1. Set a specific internal ringtone for a user's phone.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the user's extension.
   b. Click **Phone** tab to edit the phone associated with the extension.
   c. In the **Distinctive Ringtone** section, click **Add**.
   d. In the **Alert Info** field, select the alert info that is pre-defined for internal calls.

      In this example, select `Internal`.
   e. In the **Ringtone** field, select a ringtone for the internal calls.

      In this example, select `Ring1.wav`.

> **Note:**
> The available ringtones vary by phone models.



f. Click **Save**.

2. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to the user's extension.

Result

The user's phone plays ringtone *Ring1.wav* when receiving internal calls.

Set Distinctive Ringtones for External Calls

You can set distinctive ringtones on different inbound routes. When an extension user hears the ringtone of an external incoming call, the user may notice the intention of the call.

Procedure

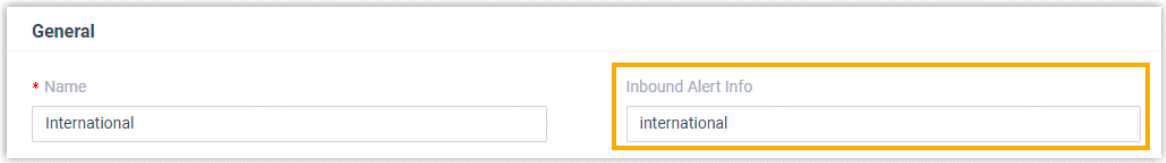
1. Set alert info for external calls on the PBX.
2. Set a specific ringtone for a phone.

Set alert info for external calls on the PBX

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, edit a desired inbound route.
2. In the **General** section, enter an alert info in the **Inbound Alert Info** field.

   The alert info is used to trigger IP phones to play a specific ringtone when receiving external calls from the inbound route.

   In this example, set alert info to `international` to identify international calls.



3. Click **Save** and **Apply**.

Set a specific ringtone for a phone

For users who want to play a specific ringtone for external calls on their phones, you can set a specific tone for their extensions by auto provisioning.

> **Note:**
> Users can also log in to phone web interface to set distinctive ringtone manually on their own IP phones. For more information, contact the phone manufacturer.

Prerequisites

The user's extension should have been associated with a phone.

For more information, see Auto Provision IP Phones.

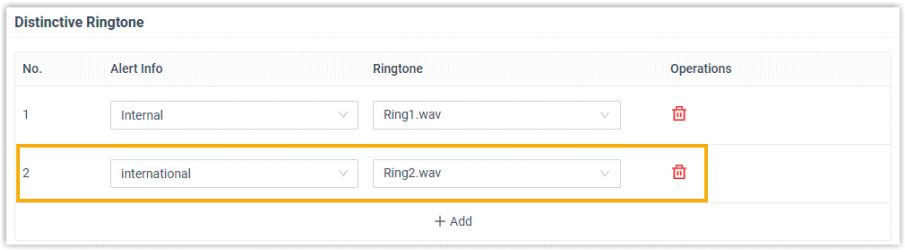
Procedure

1. Set distinctive ringtones for a user.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the user's extension.
   b. Click the **Phone** tab.
   c. In the **Distinctive Ringtone** section, click **Add**.
   d. In the **Alert Info** field, select the alert info that is pre-defined for external calls.

      In this example, select `international`.
   e. In the **Ringtone** field, select a specific ringtone for international calls.

      In this example, select `Ring2.wav`.

      > **Note:**
      > The available ringtones vary by phone models.



f. Click **Save**.

2. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to user's extension.

Result

The user's phone plays ringtone *Ring2.wav* when receiving external calls from the specific inbound route.

Set Distinctive Ringtones for Queue Calls

You can set a unique ring tone per call queue so that the agents can easily identify who is calling. This is especially useful for the agents who are in multiple call queues to help them identify calls.

Procedure

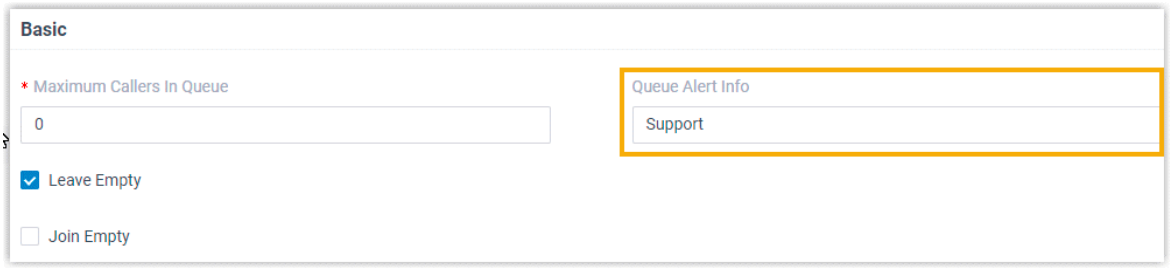
1. Set an alert info for queue calls on the PBX.
2. Set a specific ring tone for a phone.

Set an alert info for queue calls on the PBX

1. Log in to PBX web portal, go to **Call Features > Queue**, edit a desired queue.
2. Click the **Preferences** tab.
3. In the **Basic** section, enter an alert info in the **Queue Alert Info** field.

   The alert info is used to trigger IP phones to play a specific ring tone when receiving a call through this queue.

   In this example, set the alert info to `Support`.



Set a specific ring tone for a phone

For the agents who want to play unique ring tones for different queue calls on their phones, you can set distinctive ring tones for their phones by auto provisioning.

> **Note:**
> Users can also log in to phone web interface to set distinctive ring tone manually on their own IP phones. For more information, contact the phone manufacturer.

Prerequisites

The agent's extension should have been associated with a phone.

For more information, see Auto Provision IP Phones.

Procedure

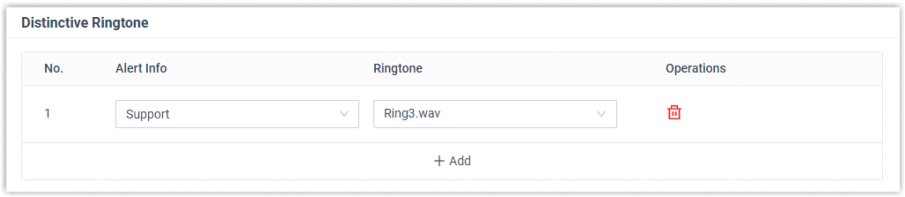
1. Set a specific queue ring tone for an agent's phone.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the agent's extension.
   b. Click the **Phone** tab.
   c. In the **Distinctive Ringtone** section, click **Add**.
   d. In the **Alert Info** field, select the alert info that is pre-defined for queue calls.

      In this example, select `Support`.
   e. In the **Ringtone** field, select a specific ring tone for the queue calls.

      In this example, select `Ring3.wav`.

      > **Note:**

> The available ring tones vary by phone models.



f. Click **Save**.

2. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to agent's extension.

Result

The agent's phone plays ringtone *Ring3.wav* when receiving calls from the Support queue.

Set Distinctive Ringtones for Ring Group Calls

You can set a unique ringtone per ring group so that the members can easily identify who is calling. This is especially useful for the members who are in multiple ring groups to help them identify calls.

Procedure

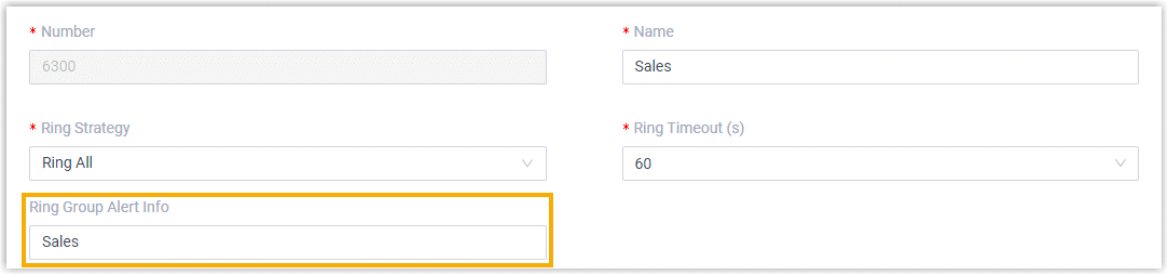
1. Set an alert info for ring group calls on the PBX.
2. Set a specific ringtone for a phone.

Set an alert info for ring group calls on the PBX

1. Log in to PBX web portal, go to **Call Features > Ring Group**, edit a desired ring group.
2. In the **Ring Group Alert Info** section, enter an alert info.

   The alert info is used to trigger IP phones to play a specific ringtone when receiving a call through this ring group.

   In this example, set alert info to `Sales`.



Set a specific ringtone for a phone

For ring group members who want to play a specific ringtone for ring group calls on their phones, you can set a specific tone for their extensions by auto provisioning.

> **Note:**
> Users can also log in to phone web interface to set distinctive ringtone manually on their own IP phones. For more information, contact the phone manufacturer.

Prerequisites

The ring group member's extension should have been associated with a phone.

For more information, see Auto Provision IP Phones.

Procedure

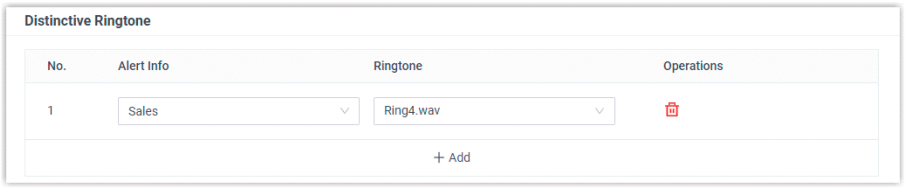
1. Set a specific ringtone for a ring group member.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit ring group member's extension.
   b. Click the **Phone** tab.
   c. In the **Distinctive Ringtone** section, click **Add**.
   d. In the **Alert Info** field, select the alert info that is pre-defined for ring group calls.

      In this example, we select `Sales`.
   e. In the **Ringtone** field, select a specific ringtone for the ring group calls.

      In this example, we select `Ring4.wav`.

      > **Note:**

> The available ringtones vary by phone models.



f. Click **Save**.

2. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to ring group member's extension.

Result

The ring group member's phone plays ringtone *Ring4.wav* when receiving calls from the Sales ring group.

Set Distinctive Ringtones for IVR Calls

You can set a unique ringtone per IVR so that the extension users can easily identify who is calling.

Procedure

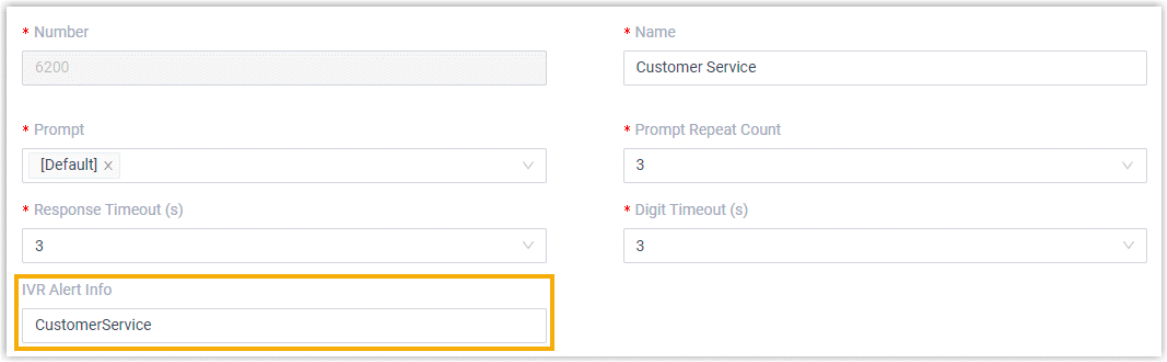
1. Set an alert info for IVR calls on the PBX.
2. Set a specific ringtone for a phone.

Set an alert info for IVR calls on the PBX

1. Log in to PBX web portal, go to **Call Features > IVR**.
2. In the **IVR Alert Info** field, enter an alert info.

   The alert info is used to trigger IP phones to play a specific ringtone when receiving a call through this IVR.

   In this example, set alert info to `CustomerService`.



Set a specific ringtone for a phone

For users who want to play a specific ringtone for IVR calls on their phones, you can set a specific tone for their extensions by auto provisioning.

> **Note:**
> Users can also log in to phone web interface to set distinctive ringtone manually on their own IP phones. For more information, contact the phone manufacturer.

Prerequisites

The user's extension should have been associated with a phone.

For more information, see Auto Provision IP Phones.

Procedure

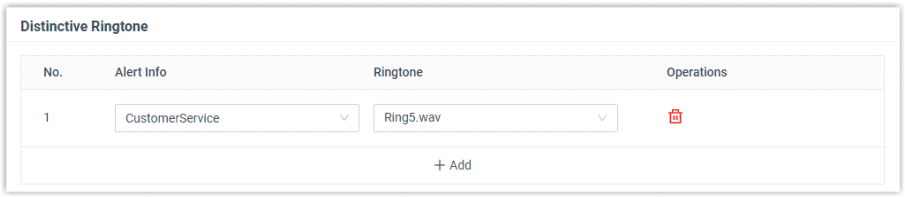
1. Set a specific ringtone for a user.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the user's extension.
   b. Click the **Phone** tab.
   c. In the **Distinctive Ringtone** section, click **Add**.
   d. In the **Alert Info** field, select the alert info that is pre-defined for IVR calls.

      In this example, select `CustomerService`.
   e. In the **Ringtone** field, select a specific ringtone for the IVR calls.

      In this example, select `Ring5.wav`.

      > **Note:**

> The available ringtones vary by phone models.



f. Click **Save**.

2. Reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to user's extension.

Result

The user's phone plays ringtone *Ring5.wav* when receiving calls from CustomerService IVR.

Distinctive Caller ID Name

Distinctive Caller ID Name Overview

This topic describes what is Distinctive Caller ID Name, and an example of Distinctive Caller ID Name.

What is Distinctive Caller ID Name

Distinctive Caller ID Name allows employees to know where the incoming call is routed, and who is calling. Distinctive Caller ID Name is a string that will be displayed on employees' phones, which may include the followings (from the highest to the lowest):

1. Contact name that is stored in Company Contacts directory or Personal Contacts directory.

   > **Note:**
   > If the extension user does not have permission to view Company Contacts, the contact name stored in Company Contact will not be displayed on the extension user's phone.

2. Call feature name (the name of IVR, Ring Group, or Queue)

> **Note:**
> If an incoming call reaches an extension through multiple call features, the name of the last call feature will be displayed. For example, if a call reaches an IVR and then goes to a queue, the queue name will be displayed on agents' phones.

3. Trunk DID/DDI name
4. Caller Name (CNAM): CNAM is sent from the caller that displays the caller name or the caller's company name.

> **Note:**
> CNAM is configured by the caller's side.

An example of Distinctive Caller ID Name

Your company has a support team that is responsible for providing technical services for customers from China and America. The following settings are configured on your PBX to achieve your goal:

Queue

A queue named "Support" for support team.

SIP trunk

A SIP trunk with two DID numbers that are bound with their respective names.

Table 27.

DID Number	DID Name
1258888	China
1256666	America

Assume that you have the following two contacts stored in your Company Contact directory.

Name	Phone Number
Sunmy	5502222
Becky	5503333

When customers dial different DID numbers and reach the Support queue, the caller ID names displayed differently on agents' phones. The display priority of Distinctive Caller ID Name is as below:

{contact_name}: *{queue_name}*: *{trunk_did_name}*: *{caller_name}*

> **Note:**
> If none of the above names are provided, the names will not be displayed.

Example:

- Customer Becky dials 1258888 to reach the Support team and no Caller Name is sent from Becky, the caller ID name displayed is *Becky: Support: China*.
- Customer Sunmy dials 1256666 to reach the Support team and no Caller Name is sent from Sunmy, the caller ID name displayed is *Sunmy: Support: America*.
- Customer C dials 1258888 to reach the Support team and a Caller Name "Yeastar" is sent from customer C, the caller ID name displayed is *Support: China: Yeastar*

Enable or Disable Distinctive Caller ID Name

You can decide whether to display a call feature name (Queue name, IVR name, or Ring Group name) or a name associated with a trunk DID/DDI number when an incoming call reaches.

Enable or disable the display of call feature name

The call feature name refers to the name of an IVR, a Ring Group, or a Queue.

1. Log in to PBX web portal, go to **PBX Settings > > Preferences**.
2. In the **Distinctive Caller ID Name** section, configure the followings:
    - To display Queue names, Ring Group names, and IVR names, select the check-box of **Display Call Feature Name**.
    - To hide Queue names, Ring Group names, and IVR names, unselect the check-box of **Display Call Feature Name**.



3. Click **Save** and **Apply**.

Enable or disable the display of trunk DID/DDI name

1. Log in to PBX web portal, go to **PBX Settings > Preferences**.
2. In the **Distinctive Caller ID Name** section, configure the followings:
   - To display trunk DID/DDI name, select the checkbox of **Display DID/DDI Name**.
   - To hide trunk DID/DDI name, unselect the checkbox of **Display DID/DDI Name**.



3. Click **Save** and **Apply**.

Call Features

Voicemail

Voicemail Overview

Yeastar P-Series Cloud Edition integrates a free voicemail system. This topic describes the voicemail types, voicemail usages, voicemail personalization, and the adjustable voicemail capacity and limitations.

Voicemail types

Yeastar P-Series Cloud Edition provides two types of voicemail:

- **Extension Voicemail**: Voicemail for individual extension.
- **Group Voicemail**: Group Voicemail is a feature for a team to share the workload of reading and responding to voicemail messages.

  Group Voicemail is useful if your company is organized into departments. For example, after setting up a group voicemail for Support team, a customer can deliver voicemail messages to the Support team, then any team members can access the group voicemail box to check the customer's voicemail.

Voicemail usages

A flexible call route system for forwarding calls to voicemail:

- **Extension**: Allow the caller to leave a message when the extension user is unavailable to take a call.

  For more information, see Forward extension users' calls to voicemail.
- **Ring Group/Queue**: Failover to group voicemail if no agents are available or timeout is reached.

  For more information, see Set failover destination to voicemail for a ring group or queue.
- **IVR**: Give the customers an option to leave a voicemail message. When the customers cannot get the information from IVR, they can leave a message.

  For more information, see Allow users to leave voicemail messages by IVR.

- **Any inbound calls**: Provide a dedicated line to collect user feedback if immediate phone support is not required.

  For more information, see Forward inbound calls to voicemail.

Voicemail personalization

Various options are available for personalizing voicemail:

- **Voicemail greeting**: Custom greeting is available for global or a specific extension. The extension users can also customize their greetings based on presence.

  For more information, see Voicemail Greeting Overview.

- **Voicemail notification**: Various ways to get notified of new voicemail messages, including on IP phones, emails, or Linkus clients.

  For more information, see Voicemail Notification Overview.

- **Envelope playback**: Play optional envelope information before listening to voicemail message, including date and time, caller ID, and message duration.

  For more information, see Configure Message Envelope.
- **Caller experience**: User-friendly experience in leaving a message, such as allow the caller to review message, send a message without ringing extensions, break out of voicemail to operator, etc.
  For more information, see:
  - Allow Callers to Press a Key to Leave Messages
  - Allow Callers to Dial Extension from Voicemail
  - Allow Callers to Break out from Voicemail
  - Allow Callers to Review Voicemail Messages

Voicemail capacity and limitations

The default and adjustable capacity and limitations for each voicemail box are as follows:

- Message length: 1 to 15 minutes.

  The default minimum duration of a message is 2 seconds; the default maximum duration of a message is 10 minutes.

  To change the message length, see Limit Voicemail Message Length.
- Mailbox capacity: 1 to 500.

  The default max number of voicemail is 100.

To change mailbox capacity, see Auto Cleanup Voicemail Messages.

- Storage time: Unlimited.

  The default is 0, which means no limit.

  To change the storage time, see Auto Cleanup Voicemail Messages.

Group Voicemail

Set up Group Voicemail for a Queue

You can set up a Group Voicemail for a queue. All agents of the queue will get notified when a group voicemail message is received.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**.
2. Click **Add**.
3. Under **General** tab, complete the general settings:
   a. In the **Basic** section, configure the following settings.

Setting	Description
Type	Select **Queue**.
Queue	Select a queue.
Number	The Group Voicemail number is the queue number, and is not editable.
Name	The Group Voicemail name is the queue name, and is not editable.
Mode	Select the mode to handle received voicemail messages. • **Shared by Members**: The voicemail messages are saved in the group mailbox, and are shared by all members. Any members can manage the group voicemail messages. • **Broadcast to Members**: The voicemail messages are not stored in the group mailbox. Instead, the system broadcasts (copies and forwards) the voicemail messages to the individual mailboxes of all the members.
	Enable or disable voicemail PIN authentication.

Setting	Description
Voicemail PIN Authentication	
Voicemail Access PIN	If enable voicemail PIN authentication, enter a desired access PIN number.
Play Date and Time	Optional. Play the date and time that the message was received before a voicemail message is played.
Play Caller ID	Optional. Play the caller ID information before a voicemail message is played.
Play Message Duration	Optional. Play the duration of the message before a voicemail message is played.

b. In the **Members** section, all the agents of the queue are selected, and the members are not editable.

> **Note:**
> If the queue agents are changed, the members of the group's voice mailboxes also change.

c. In the **Group Voicemail Greeting** section, select a voicemail greeting.

You can also click **Greeting Management** to customize a greeting or mange your custom greetings.

4. Under **Voicemail Announcement** tab, set up call alerts for new voicemail messages in the group voicemail.

> **Note:**
> This feature is only supported on the group voicemail with the **Mode** set to **Shared by Members**.

5. Click **Save** and **Apply**.

Related information

Enable or Disable Voicemail Access PIN
Change Voicemail Access PIN
Configure Message Envelope
Change Voicemail Greetings
Record or Upload Voicemail Greetings
Manage Group Voicemail Greetings

Set up Group Voicemail for a Ring Group

You can set up a Group Voicemail for a ring group. All members of the ring group will get notified when a group voicemail message is received.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**.
2. Click **Add**.
3. Under **General** tab, complete the general settings:
    a. In the **Basic** section, configure the following settings:

Setting	Description
Type	Select **Ring Group**.
Ring Group	Select a ring group.
Number	The group voicemail number is the ring group number, and is not editable.
Name	The group voicemail name is the ring group name, and is not editable.
Mode	Select the mode to handle received voicemail messages. • **Shared by Members**: The voicemail messages are saved in the group mailbox, and are shared by all members. Any members can manage the group voicemail messages. • **Broadcast to Members**: The voicemail messages are not stored in the group mailbox. Instead, the system broadcasts (copies and forwards) the voicemail messages to the individual mailboxes of all the members.
Voicemail PIN Authentication	Enable or disable voicemail PIN authentication.
Voicemail Access PIN	If enable voicemail PIN authentication, enter a desired access PIN number.
Play Date and Time	Optional. Play the date and time that the message was received before a voicemail message is played.
Play Caller ID	Optional. Play the caller ID information before a voicemail message is played.
Play Message Duration	Optional. Play the duration of the message before a voicemail message is played.

b. In the **Members** section, all the members of the ring group are selected, and the members are not editable.

> **Note:**
> If the ring group members are changed, the members of the group's voice mailboxes also change.

c. In the **Group Voicemail Greeting** section, select a voicemail greeting.

You can also click **Greeting Management** to customize a greeting or mange your custom greetings.

4. Under **Voicemail Announcement** tab, set up call alerts for new voicemail messages in the group voicemail.

> **Note:**
> This feature is only supported on the group voicemail with the **Mode** set to **Shared by Members**.

5. Click **Save** and **Apply**.

Related information

Enable or Disable Voicemail Access PIN
Change Voicemail Access PIN
Configure Message Envelope
Change Voicemail Greetings
Record or Upload Voicemail Greetings
Manage Group Voicemail Greetings

Set up Group Voicemail for a Custom Group

For a team whose members come from different departments, you can set up a Group Voicemail for the team members. All team members will get notified when a group voicemail message is received.

Procedure

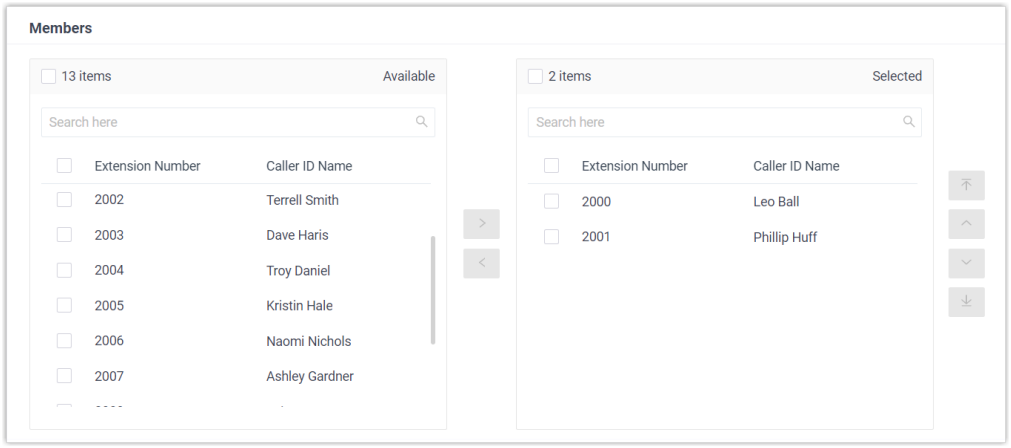
1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**.
2. Click **Add**.
3. Under **General** tab, complete the general settings:

a. In the **Basic** section, configure the following settings:

Setting	Description
Type	Select **Custom**.
Number	Specify a virtual number for callers to access the group voicemail.
Name	Enter a group voicemail name to help you identify it.
Mode	Select the mode to handle received voicemail messages. • **Shared by Members**: The voicemail messages are saved in the group mailbox, and are shared by all members. Any members can manage the group voicemail messages. • **Broadcast to Members**: The voicemail messages are not stored in the group mailbox. Instead, the system broadcasts (copies and forwards) the voicemail messages to the individual mailboxes of all the members.
Voicemail PIN Authentication	Enable or disable voicemail PIN authentication.
Voicemail Access PIN	If enable voicemail PIN authentication, enter a desired access PIN number.
Play Date and Time	Optional. Play the date and time that the message was received before a voicemail message is played.
Play Caller ID	Optional. Play the caller ID information before a voicemail message is played.
Play Message Duration	Optional. Play the duration of the message before a voicemail message is played.

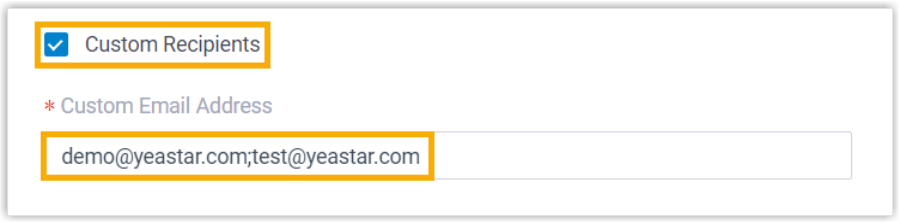
b. In the **Members** section, set the recipients to receive group voicemails.

- To send group voicemails to extension users, select the desired extensions or extension groups from **Available** box to **Selected** box.



- To send group voicemails to recipients without extension assigned, select the checkbox of **Custom Recipients**, and enter email addresses in the **Custom Email Address** field.

> **Note:**
> A maximum of 5 email addresses is supported; Use semicolons ; to separate multiple email addresses.



c. In the **Group Voicemail Greeting** section, select a voicemail greeting.

You can also click **Greeting Management** to customize a greeting or mange your custom greetings.

4. Under **Voicemail Announcement** tab, set up call alerts for new voicemail messages in the group voicemail.

> **Note:**
> This feature is only supported on the group voicemail with the **Mode** set to **Shared by Members**.

5. Click **Save** and **Apply**.

Related information

Enable or Disable Voicemail Access PIN
Change Voicemail Access PIN
Configure Message Envelope
Change Voicemail Greetings
Record or Upload Voicemail Greetings
Manage Group Voicemail Greetings

Set up Call Alerts for Group Voicemail

Yeastar P-Series Cloud Edition allows you to configure call alerts for new group voicemail messages. When a new voicemail message is left, the system will call the specified phone numbers in a pre-defined priority order until all the voicemail messages are read or the specified call cycles are reached.

Scenario

A company has a team of on-call agents responsible for after-hours emergency. Instead of answering incoming calls, the company wants the calls to be routed to voicemail while the agents can be alerted as soon as possible. In this case, the company can configure an inbound route for such emergency calls, routing the calls to a dedicated group voicemail, and then set up call alerts to notify the agents whenever a new voicemail message is left.

Requirements and restrictions

Before you begin, read through the requirements and restrictions for the feature:

Requirements

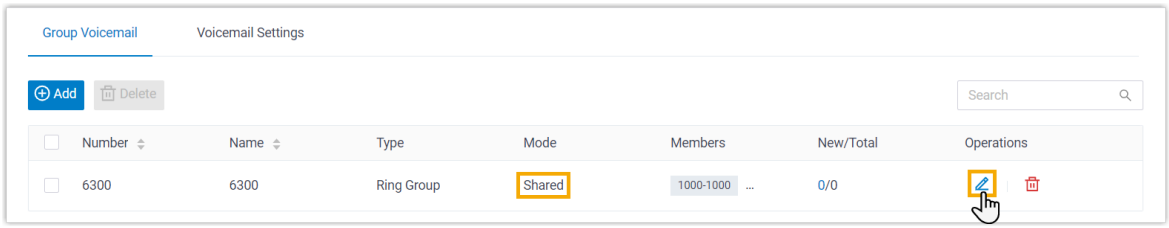
- **Version**: 84.13.0.25 or later
- **Plan**: Enterprise Plan or Ultimate Plan

Restrictions

- This feature is only available for the group voicemails that are **Shared by Members**.
- Allow calls to only **External Number** when there is a new voicemail message and only **5** external numbers are supported.

Procedure

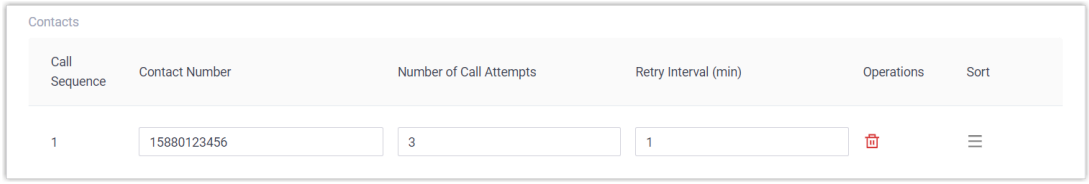
1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit a group voicemail with the **Mode** displayed as **Shared**.



2. Click the **Voicemail Announcement** tab.
3. Turn on call notifications for new group voicemail messages, and set up the number and rule to call.
    a. Turn on the switch of **Voicemail Announcement**.
    b. In the **Max Number of Call Cycles** field, specify how many times you want to repeatedly call the contact list.

> **Note:**
> - The value you enter should not be greater than 30.
> - Notification calls will continue until the number of call cycle is reached or all voicemail messages have been read.

    c. In the **Contacts** section, click **Add** to configure the number to call and set up the rule.



- **Contact Number**: Enter an external number.

> **Note:**
> Notification calls to the number can be made via any matched outbound routes. Make sure that there are outbound routes that match the number, otherwise the notification call cannot be made.

- **Number of Call Attempts**: Specify the number of attempts to call the number if the callee doesn't answer the call or if the callee answers the call but does not listen to the new voicemail message.
- **Retry Interval (min)**: Specify the interval between the call attempts.

d. **Optional:** To add more contacts, repeat step c.

> **Note:**
> The system will call the external numbers from the top down. You can adjust the priority order using ☰.

4. Click **Save**.

Result

When the group voicemail receives a new message, the PBX will call the preset numbers in sequence. The callee can answer the call and follow the prompt to listen to the message or call the original caller back directly.

Related information

Extension Call Statistics Report
Extension Call Activity Report

Manage Group Voicemails

This topic describes how to edit a group voicemail, and delete group voicemails.

Edit a group voicemail

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**.
2. Click ✎ beside the group voicemail that you want to edit.
3. Change the settings according to your needs.
   - Enable or Disable Voicemail Access PIN
   - Change Voicemail Access PIN
   - Configure Message Envelope
   - Change Voicemail Greetings
   - Record or Upload Voicemail Greetings
   - Manage Group Voicemail Greetings
4. Click **Save** and **Apply**.

Delete group voicemails

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**.
2. To delete a group voicemail:
   a. Click  beside the group voicemail that you want to delete.
   b. Click **Apply**.
3. To delete group voicemails in bulk:
   a. Select the checkboxes of the group voicemails that you want to delete, click **Delete**.
   b. Click **OK** and **Apply**.

Send and Receive Voicemail Messages

Forward Calls to Voicemail

Never miss a lead by allowing your customers to leave voicemail messages. This topic describes how to forward various kinds of calls to voicemail.

Background information

A growing business cannot afford to miss incoming calls. A missed call may make your customers impatient. Forwarding calls to voicemail automatically helps you to stay connected with customers and enhance the service.

In the following scenarios, you can consider a destination as voicemail, which helps the system to forward calls to voicemail:

- Forward extension users' calls to voicemail: The extension user is unavailable to answer a call.
- Set failover destination to voicemail for a ring group or queue: No members or agents are available to take a call or the call reaches the timeout.
- Allow users to leave voicemail messages by IVR: Give the customers an option to leave a voicemail message. When the customers cannot get the information from IVR, they can leave a message.
- Forward inbound calls to voicemail: Immediate phone support is not required.

Forward extension users' calls to voicemail

You can set call forwarding rules for each presence status as users' need, the system will forward extension users' calls to voicemail according to the presence status.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Presence** tab, select a presence status to configure.
3. In the **Call Forwarding** section, configure call forwarding rules for internal calls (incoming calls from colleagues) and external calls (inbound calls from customers).
    a. Select the checkbox of a forwarding condition.
    b. Select a corresponding destination for the forwarding condition to one of the following options:
        - **Voicemail**: Forward calls to the extension's voicemail box.
        - **Group Voicemail**: Forward calls to a selected group mailbox.
4. Click **Save** and **Apply**.

Set failover destination to voicemail for a ring group or queue

1. Log in to PBX web portal, set failover destination to voicemail.
    - To set failover destination for a ring group, go to **Call Features > Ring Group**, edit the desired ring group.
    - To set failover destination for a queue, go to **Call Features > Queue**, edit the desired queue.
2. In the **Failover Destination** drop-down list, select a corresponding destination to one of the following options:
    - **Extension Voicemail**: Forward calls to the extension's voicemail box.
    - **Group Voicemail**: Forward calls to a selected group mailbox.
3. Click **Save** and **Apply**.

Allow users to leave voicemail messages by IVR

Prerequisites

Update your IVR prompt that would instruct callers to press a key to access voicemail.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. Click **Key Press Event** tab, select a corresponding destination to one of the following options:
    - **Extension Voicemail**: Forward calls to the extension's voicemail box.

- **Group Voicemail**: Forward calls to a selected group mailbox.

3. Click **Save** and **Apply**.

Forward inbound calls to voicemail

On non-working days, you can forward inbound calls to voicemail.

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, edit the desired inbound route.
2. In the **Default Destination** section, select a corresponding destination to one of the following options:
   - **Extension Voicemail**: Forward calls to the extension's voicemail box.
   - **Group Voicemail**: Forward calls to a selected group mailbox.
3. Click **Save** and **Apply**.

Leave a Voicemail Message without Calling the User

This topic describes how to send a voicemail message without ringing extensions.

Background information

Although you can send a message by email or text, sometimes there is no replacement for the emotion, inflection, and sincerity of your voice. When you do not want to disturb someone or when you do not have time for a phone conversation, you can send a voicemail message without calling extension.

It is useful in a team work. When your partners are busy in a meeting or after work, but you have some information that need to share with them, you can send a voicemail message without calling them. It allows your partner to reflect prior to responding.

Prerequisites

This feature is only for internal extension users.

Procedure

1. To leave a voicemail message to a specific extension user, dial feature code (`*12`) followed by extension number (for example, *121001).
2. To leave a voicemail message to a queue, a ring group, or a custom group, dial feature code (`*12`) followed by group voicemail number (for example, *126100).
3. Follow the voice prompt to leave your message.
4. When done, hang up or press `#`.

> **Tip:**
> The default feature code for sending voicemail messages is `*12`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Voicemail > Leave a Voicemail for Extension/Group Voicemail**.

Forward Voicemail Messages to Email

Email is one of the most popular communication tools for business. Forwarding voicemail messages to email is an efficient business feature that allows employees to receive voicemail audio files as email attachments. This topic describes what you can do with voicemail to email and how to forward voicemail message to email for specific extension users.

Background information

Scenario

For employees who travel frequently and require an efficient way to keep up with voicemail and provide a quick response for the customers, it is an efficient way to get alert timely, listen to voicemails anywhere, and handle business over email.

Benefit

Each time the employees receive a voicemail message, they can receive an email with the new voicemail message attached as a .wav file, including caller ID, time of the call, and callback number.

- **Easy to identify**: In emails, the employees can quickly identify the person who left the message, and listen to voicemail message as they need.
- **Easy to listen**: The employees can check and listen to their voicemail messages via computer, smart phone or mobile device at convenience, instead of calling to voicemail box and navigating through the maze of voice prompts. They can also fast-forward or rewind to reach and repeat the important portion.
- **Easy to share**: The employees can forward emails to share voice messages with teammates to improve collaboration efficiency.
- **Easy to manage**: Managing the communications is easier since all the voicemail messages are in the email box. It is faster to sort, prioritize, scan, delete, and save voicemail message.

Prerequisites

- Make sure there is a valid email address assigned to each extension.
- Make sure the PBX system email works, or the PBX cannot forward the received voicemail to an extension user's email.

Procedure

1. Log in to PBX web portal, go to **Extension and trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **New Voicemail Notification** drop-down list, select **Send Email Notifications with Attachment**.
4. In the **After Notification** drop-down list, set how to handle the voicemail message after the system has successfully notified the extension user by email.
5. Click **Save** and **Apply**.

Manage Voicemail Messages

Check Voicemail Messages

This topic describes how to check voicemail messages.

Background information

Methods

Extension users can get an instant voicemail notification when receiving a new voicemail message. There are multiple ways to check voicemail messages anytime and anywhere.

- On an IP phone
- On Linkus client
- Via Email
- Via IVR

Feature code

The default feature code for checking voicemail messages is *2. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Voicemail > Check Voicemail/Subscribe Voicemail Status**.

Check voicemail messages on an IP phone

Check voicemail messages on a user's own phone

1. Dial feature code *2.
2. Follow the voice prompt to enter your PIN number followed by #.
3. Navigate through the voicemail menu to check your voicemail message.

Check voicemail messages from another phone

1. Dial feature code *2 followed by the extension number whose voicemail will be checked. (for example, to check voicemail of extension 1001, dial *21001).
2. Follow the voice prompt to enter your PIN number followed by #.
3. Navigate through the voicemail menu to check your voicemail message.

Check group voicemail messages from an IP phone

If the **Mode** of group voicemail is set to **Shared by Members**, the users can check the messages in group mailbox. If any users check the new messages, the status of messages will be set as read.

1. Dial feature code *2 followed by the group voicemail number (for example, *26100).
2. Navigate through the voicemail menu to check your voicemail message.

Check voicemail messages on Linkus client

If you have enabled **Linkus Clients** for extension users, the extension users can check voicemail messages on their Linkus clients.

Check voicemail messages via Email

If you have set up the feature of forwarding voicemail messages to email for extensions, the extension users can check their voicemail messages in their email boxes.

Check voicemail messages via IVR

If you have allowed extension users to dial in an IVR to check voicemail messages, the extension users can also check voicemail messages when they are out of office.

1. Dial in an IVR, follow the voice prompt.
2. Dial feature code *2 followed by extension number or group voicemail number, and then enter the PIN number.
3. Navigate through the voicemail menu to check voicemail messages.

Enable or Disable Voicemail Transcription

Yeastar P-Series Cloud Edition supports a Voicemail Transcription feature. Using this feature can transcribe voice messages to texts, users can view the message content directly, which brings great convenient and efficiency.

Enable Voicemail Transcription

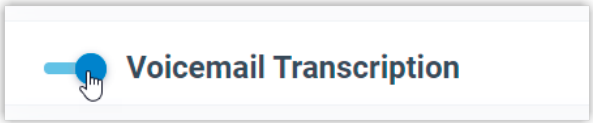
Prerequisites

Voicemail Transcription feature requires the use of a third-party transcription service to convert the voice message to text. Before you start to use Voicemail Transcription, make sure that the PBX is integrated with a third-party Speech-to-Text (STT) service.

For now, Yeastar P-Series Cloud Edition allows you to integrate with Google Cloud STT API service. For more information, see Integrate Yeastar P-Series Cloud Edition with Google Cloud Speech-to-Text Service.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Voicemail Settings**.
2. Scroll down to the bottom of the page, turn on **Voicemail Transcription**.



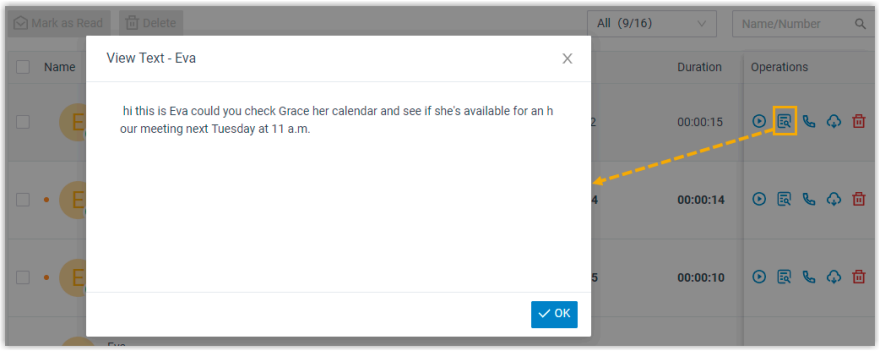
3. Click **Save**.

Result

The Voicemial Transcription feature is enabled, users can receive voicemails in the form of text on different platforms.

Linkus UC Clients

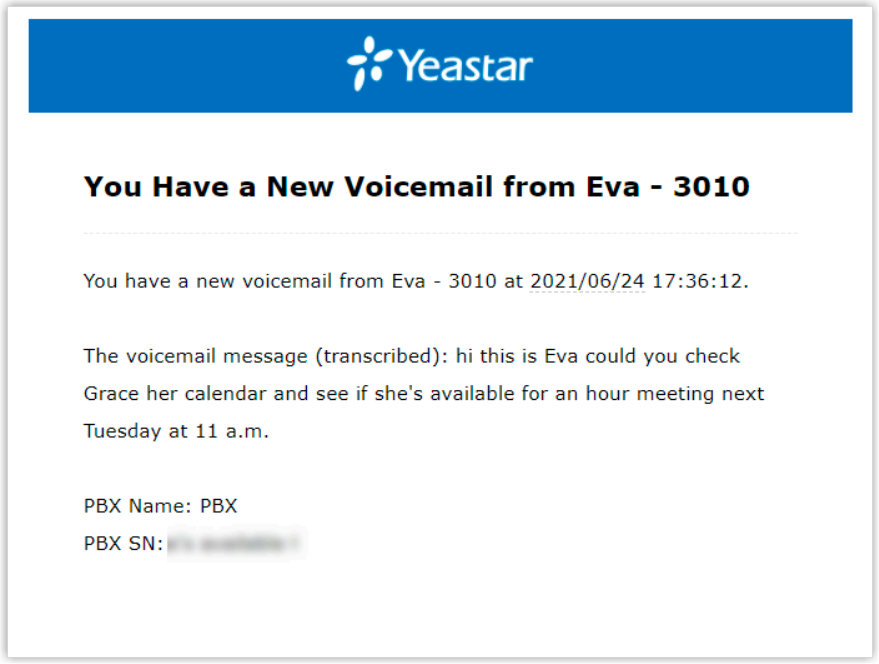
Users can check the transcribed text for each voicemail on Linkus Web Client, Desktop Client, and Mobile Client.



Email Client

If Voicemail to Email feature is enabled, the transcribed text will be displayed in the email content for received voicemails.

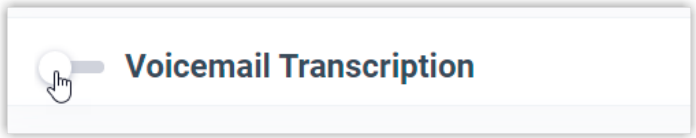
The figure below shows an example of voicemail notification email.



Disable Voicemail Transcription

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Voicemail Settings**.
2. Scroll down to the bottom of the page, turn off **Voicemail Transcription**.



3. Click **Save**.

Result

The Voicemail Transcription feature is unavailable.

Configure Message Envelope

This topic describes how to enable or disable message envelope.

Background information

Message envelope is given before a voicemail message is played. Message envelop includes the following information:

- Date and Time that the message was received.
- Caller ID information.
- Duration of the message.

You can enable or disable envelope information separately according to user needs.

Configure message envelope for extension voicemail

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. Decide whether to enable the following information for the message envelope:
    - **Play Date and Time**: Play the date and time that the message was received.

- **Play Caller ID**: Play the caller ID information.
- **Play Message Duration**: Play the duration of message.

4. Click **Save** and **Apply**.

Configure message envelope for group voicemail

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Basic** section, decide whether to enable the following information for the message envelope:
   - **Play Date and Time**: Play the date and time that the group voicemail message was received.
   - **Play Caller ID**: Play the caller ID information.
   - **Play Message Duration**: Play the duration of group voicemail message.
3. Click **Save** and **Apply**.

Limit Voicemail Message Length

Limiting voicemail message length is a good way to reduce invalid or lengthy voicemails. This topic describes how to specify the message length (max and min) for a caller to leave a voicemail message.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Voicemail Settings > Message Options**.
2. In the **Max Message Time(s)** drop-down list, select a number of seconds.

   Messages exceeding the maximum duration will be automatically cut off.
3. In the **Min Message Time(s)** drop-down list, select a number of seconds.

   Messages less than the minimal duration will be automatically discarded.
4. Click **Save** and **Apply**.

> **Tip:**
> You may need to inform the callers in the greeting to keep their messages brief or under the maximum duration.

Auto Cleanup Voicemail Messages

Clean up old messages to free up space for new voicemail messages. You can determine how many and how long that the system retains voicemail messages in a mailbox. The system automatically deletes the old voicemail messages when the threshold is reached. This topic describes how to set up auto cleanup of voicemail messages for each extension.

Procedure

1. Log in to PBX web portal, go to **System > Storage > Auto Cleanup > Voicemail Auto Cleanup**.
2. In the **Max Number of Voicemail** field, enter the maximum number of voicemail messages that should be retained for each mailbox.
3. In the **Voicemail Preservation Days** field, enter the maximum number of days that voicemail messages should be retained.

   > **Note:**
   > The value `0` indicates no limit.

4. Click **Save**.

Result

If **Auto Clean up Reminder** is enabled, and the retained voicemail messages reach 90% of the threshold, the system sends you a notification email.

Voicemail Security

Change Voicemail Access PIN

This topic describes how to change voicemail access PIN for extension voicemail and group voicemail.

Background information

By default, the extension users need to enter the voicemail access PIN for authentication when checking their voicemail messages. The default voicemail access PIN is randomly generated.

Note:
The PIN can be numerics only, and a minimum of 3 digits is required.

Change voicemail access PIN for extension voicemail

There are two ways to change voicemail access PIN:

- On web interface
- Via voicemail mailbox

Change extension voicemail access PIN on web interface

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Access PIN** field, enter a PIN number.
4. Click **Save** and **Apply**.

Change extension voicemail access PIN via voicemail mailbox

1. Dial **2* to enter mailbox, enter the access PIN.
2. Press *4* to change the voicemail access PIN.
3. Follow the voice prompt, and enter the new PIN followed by *#* key.

   The call ends automatically after saving the new PIN.

Change voicemail access PIN for group voicemail

There are two ways to change voicemail access PIN:

- On web interface
- Via voicemail mailbox

Change group voicemail access PIN on web interface

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Voicemail Access PIN** field, enter a PIN number.
3. Click **Save** and **Apply**.

Change group voicemail access PIN via voicemail mailbox

1. Dial **2* followed by the group voicemail number to enter mailbox, enter the access PIN.
2. Press *4* to change the voicemail access PIN.
3. Follow the voice prompt, and enter the new PIN followed by *#* key.

   The call ends automatically after saving the new PIN.

Enable or Disable Voicemail Access PIN

A voicemail access PIN is helpful to prevent unauthorized access. This topic describes how to enable or disable voicemail access PIN.

> **Note:**
> For security reasons, we recommend that you enable voicemail access PIN.

Enable or disable voicemail access PIN for extension voicemail

1. Log in to PBX web portal, go to **Extension and Trunk > Extensions**, edit the desired extension.
2. Click **Voicemail** tab.
3. To enable voicemail access PIN, select **Enabled** from the **Voicemail PIN Authentication** drop-down list.
4. To disable voicemail access PIN, select **Disabled** from the **Voicemail PIN Authentication** drop-down list.
5. Click **Save** and **Apply**.

Enable or disable voicemail access PIN for group voicemail

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. To enable voicemail access PIN, select **Enabled** from the **Voicemail PIN Authentication** drop-down list.
3. To disable voicemail access PIN, select **Disabled** from the **Voicemail PIN Authentication** drop-down list.
4. Click **Save** and **Apply**.

Voicemail Greetings

Voicemail Greeting Overview

Voicemail greeting is a short message that is played before a caller records a message. Via the greeting, you can inform the callers your information, like when you will be available, other methods to contact you, or other options that the caller can use to receive assistance.

Greeting types

There are two types of voicemail greetings that you can set up for extension voicemail and group voicemail:

- **System Global Greeting**: A greeting that is applied to extension voicemail or group voicemail by default.
- **Custom Greeting**: A greeting that is personalized.

Personal greeting based on presence

For extension voicemail, extension users can choose how to play greetings in different presence:

- **Default greeting**: Play a greeting for any presence that doesn't have a personal greeting.
- **Presence greetings**: Play a personal greeting for each presence (available, away, do not disturb, lunch break, business trip, and off work).

  For example, an extension user has different greetings for Lunch Break status and Away status.

  - Lunch Break: "I’m currently on a lunch and unable to take your call".
  - Away: "I’m currently away from my desk".

Record or Upload Voicemail Greetings

This topic describes how to record or upload voicemail greetings for extension voicemail or group voicemail.

Background information

The personalized greetings can delight the callers, and let them know why you’re unavailable and how they can contact you.

Up to ten individual greetings are customizable for each voicemail. It is easy to customize greetings in two ways:

- **Upload an audio file**: Prepare an audio file.

  > **Note:**
  > The uploaded file should meet the audio file requirements.

- **Record a voicemail greeting from a phone**: Place a call from system, the extension users can answer the call and record their voice as voicemail greetings.

Record or upload voicemail greetings for extension voicemail

The extension users may want to make their voicemails more personalized and professional depending on presence, you can set personalized voicemail greetings for each user.

Upload an extension voicemail greeting

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Greeting** section, click **Greeting Management**.
4. In the pop-up window, click **Upload**.
5. Select an audio file to upload.

   You can view and manage the greeting in **Greeting Management**.

Record an extension voicemail greeting from a phone

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In **Voicemail Greeting** section, click **Greeting Management**.
4. In the pop-up window, click **Record New Greeting** tab.
5. In the **Audio File Name** field, enter a name to help you identify it.
6. In the **Extension** drop-down list, select an extension to record a greeting.
7. Click **Save**.

   The system places a call to the selected extension.
8. Answer the call, and record greeting on the phone.

Press **#** key or hang up after recording greeting, you can view and manage the greeting in **Greeting Management** tab.

Record or upload voicemail greetings for a group voicemail

Upload a group voicemail greeting

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Group Voicemail Greeting** section, click **Greeting Management**.
3. In the pop-up window, click **Upload**.
4. Select an audio file to upload.

   You can view and manage the greeting in **Greeting Management**.

Record a group voicemail greeting from phone

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In **Group Voicemail Greeting** section, click **Greeting Management**.
3. In the pop-up window, click **Record New Greeting** tab.
4. In the **Audio File Name** field, enter a name to help you identify it.
5. In the **Extension** drop-down list, select an extension to record a greeting.
6. Click **Save**.

   The system places a call to the selected extension.
7. Answer the call, and record greeting on the phone.

   Press **#** key or hang up after recording greeting, you can view and manage the greeting in **Greeting Management** tab.

Manage Personal Voicemail Greetings

This topic describes how you can manage an extension user's personal greeting, including playing, downloading, and deleting greetings.

Play a personal greeting

To check the uploaded greetings or recorded greetings, you can play the greeting on a phone or on web.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Greeting** section, click **Greeting Management**.
4. Select a greeting that you want to play, click .
5. In the pop-up window, choose how to play the greeting:
   - **Play on Web**: Click to play the greeting on the web directly.
   - **Play to Extension**: Play the greeting on a phone.
     a. Select an extension, and click **Play**.

     The system places a call to the extension.
     b. Pick up the call to listen to the greeting on the phone.

Download a personal greeting

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Greeting** section, click **Greeting Management**.
4. Select a greeting that you want to download, click .

Delete personal greetings

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Greeting** section, click **Greeting Management**.
4. To delete a greeting, do the following:
   a. Click beside the greeting.
   b. Click **OK** and **Apply**.
5. To delete greetings in bulk, do the following:
   a. Select the checkboxes of the greetings, click **Delete**.
   b. Click **OK** and **Apply**.

Manage Group Voicemail Greetings

This topic describes how you can manage group voicemail greetings, including playing, downloading, and deleting greetings.

Play a group voicemail greeting

To check the uploaded greetings or recorded group voicemail greetings, you can play the greeting on a phone or on web.

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Group Voicemail Greeting** section, click **Greeting Management**.
3. Select a greeting that you want to play, click .
4. In the pop-up window, choose how to play the greeting:
   - **Play on Web**: Click to play the greeting on the web directly.
   - **Play to Extension**: Play the greeting on a phone.
     a. Select an extension, and click **Play**.

     The system places a call to the extension.
     b. Pick up the call to listen to the greeting on the phone.

Download group voicemail greeting

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Group Voicemail Greeting** section, click **Greeting Management**.
3. Select a greeting that you want to download, click .

Delete group voicemail greetings

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Group Voicemail Greeting** section, click **Greeting Management**.
3. To delete a greeting, do the following:
   a. Click beside the greeting.
   b. Click **OK** and **Apply**.
4. To delete greetings in bulk, do the following:

a. Select the checkboxes of the greetings, click **Delete**.
b. Click **OK** and **Apply**.

Change Voicemail Greetings

Both the global and personalized voicemail greeting are changeable. This topic describes how to change voicemail greetings for extension voicemail and group voicemail.

Change global voicemail greetings for all voicemails

Prerequisites

Upload a custom greeting or record a custom greeting.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Voicemail Settings > Greeting Options**.
2. In the **Global Voicemail Greeting** drop-down list, select an audio prompt.
3. Click **Save** and **Apply**.

Result

The global voicemail greeting will be applied to all the extension voicemails and group voicemails that do not have a custom greeting.

Change voicemail greetings for a specific extension

Prerequisites

Record or upload voicemail greeting for the specific extension.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **Voicemail Greeting** section, select a greeting:
   - **Default Greeting**: Select a greeting from **Default Greeting** drop-down list.

     Default greeting is played for the presence with **Presence Greetings** set to **None**.

- **Presence Greetings** (**Available**, **Away**, **Do Not Disturb**, **Lunch Break**, **Business Trip**, and **Off Work**): Select a greeting from the corresponding presence drop-down list.

  The presence greeting is played based on extension presence.

> **Tip:**
> You can also select **Record New** to add a new greeting and apply.

4. Click **Save** and **Apply**.

Change voicemail greetings for a group voicemail

Prerequisites

Record or upload voicemail greeting for the group voicemail.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Group Voicemail**, edit the desired group voicemail.
2. In the **Group Voicemail Greeting** section, select a greeting.

> **Tip:**
> You can also select **Record New** to add a new greeting and apply.

3. Click **Save** and **Apply**.

Voicemail Notifications

Voicemail Notification Overview

Extension users can get an instant notification when receiving a new voicemail message. This topic describes various ways to get notified of new voicemail messages.

Notification on IP phones

There are two methods that you can use to monitor voicemail status on an IP phone.

Monitor voicemail status by function keys

You can use a function key to monitor changes of voicemail status, including monitor your voicemails, other users' voicemails, or group voicemails. It is useful when sharing a single voicemail in a team. The team members can monitor and access the voicemail in time. Once someone reads or deletes the message, no one else should have to deal with it.

For more information, see Monitor Voicemail Status on an IP Phone.

Monitor voicemail status by MWI

Message Waiting Indicator (MWI) is a commonly supported phone feature that alerts you when receiving a new voicemail message. MWI typically involves a flashing light and optional audio alert. This can differ from device to device.

Notification by email

You can set up email notification for extension users. When receiving a voicemail message, users can get alert timely, read the message at a glance to see the caller and when the message is left, and listen to voicemails. This improves work efficiency.

- For employees who do not use the phone frequently, they don't need to pay attention to keep checking voicemail on the phone at all time.
- For employees who travel frequently, they can process voice messages in real time and respond to customers promptly.

For more information, see Set up Email Notifications for Voicemail.

Monitor Voicemail Status on an IP Phone

This topic describes how to monitor voicemail status on an IP phone by function keys.

Background information

For extension users who want to monitor voicemail status on their phones, you can set a function key for each extension user by auto provisioning.

> **Note:**
> Users can also set function keys manually on their own IP phones. For more information, contact the phone manufacturer.

Procedure

1. Assign function keys for extension users to monitor voicemail status.

a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.

- If you want to assign function keys for a specific extension user, click beside the desired extension.
- If you want to assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.

b. Click the **Function Keys** tab.

c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

- **Type**: Select the voicemail type that you want to monitor.
  - To monitor extension voicemail, select **Check Voicemail**.
  - To monitor group voicemail in shared mode, select **Check Group Voicemail**.

    > **Note:**
    > Monitor voicemail by function key is not applicable for group voicemail in broadcast mode, because the voicemail messages are not stored in the group mailbox.

- **Value**: Select an extension voicemail or group voicemail that you want to monitor.
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.
3. If the extension has been associated with a phone, reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.

Result

The function key shows the real-time status of voicemail.

- **Green**: The monitored extension has no unread voicemail messages.
- **Red**: The monitored extension has unread voicemail messages.

  To check the voicemail message, press the function key to access the voicemail box and operate following by the prompt instructions.

> **Note:**
> The key LED status may vary by phone models.

Set up Email Notifications for Voicemail

This topic describes how to set up email notifications for new voicemail messages.

Limitation

This feature is only for extensions' personal voicemails. New voicemail messages to Group Voicemail doesn't support email notifications.

> **Note:**
> The group voicemail in broadcast mode will forward messages to extensions' personal voicemails, the extension users can also receive email notifications.

Prerequisites

- Make sure there is a valid email address assigned to each extension user.
- Make sure the PBX system email works, or the PBX cannot send voicemail messages to an extension user's email.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Click **Voicemail** tab.
3. In the **New Voicemail Notification** drop-down list, decide whether to send email notifications when the user receives a new voicemail.

- To disable email notifications, select **Do Not Send Email Notifications**.
- To enable email notifications, select one of the following options:
  - **Send Email Notifications with Attachment**: Send a notification email with the new voicemail message attached as a `.wav` file.
  - **Send Email Notifications without Attachment**: Send notification emails only.

4. If you enabled email notifications, configure the following settings as needed:

Setting	Description
After Notification	Decide how to deal with voicemails after notification emails are sent out. • **Mark as Read**: Mark the voicemail message in mailbox as read. • **Delete Voicemail**: Delete the voicemail messages from mailbox. • **Do Nothing**: Keep the voicemail message in mailbox as unread.
Send to	Specify the email address for receiving notification emails. • **User Email**: Send notification emails to the user's email address. • **Custom Email**: Send notification emails to a custom email address. Enter the desired email address in the **Custom Email Address** field.

5. Click **Save** and **Apply**.

Custom Voicemail Experience

Allow Callers to Press a Key to Leave Messages

This topic describes how to allow callers to press a key to leave messages.

Background information

By default, when the caller accesses a user's voicemail, PBX starts to record message automatically. It may make callers embarrassed when they are not ready to leave a message or they don't need to leave a message. Even if the caller hangs up directly, the voice mailbox still generates a lot of invalid information.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail > Voicemail Greetings**.
2. In the **Caller Options** section, select the checkbox of **Ask callers to press 5 for leaving a message**.
3. Click **Save**.

Result

The caller can choose whether to leave a message after listening to the greeting, and press 5 to leave a message after he or she is ready.

What to do next

If a custom greeting is used for voicemail, update the greeting that would instruct callers to press 5 for leaving a message.

Allow Callers to Dial Extension from Voicemail

This topic describes how to allow callers to dial extension from voicemail.

Background information

For the employees working in multiple places, they can record a greeting to prompt the caller to dial another extension to reach them. Instead of hanging up and calling again, you can allow the caller to dial extensions directly from voicemail.

It is also useful when the boss is unavailable to answer a call, instead of leaving a message in emergency, the caller can dial the secretary's extension, .

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail**.
2. In the **Caller Options** section, select the checkbox of **Allow callers to dial extension**.
3. Select the extensions that can be dialed from the **Available** box to the **Selected** box.
4. Click **Save** and **Apply**.

Result

The caller can press * key to dial an extension.

What to do next

If a custom greeting is used for voicemail, update the greeting that would instruct callers to press * key for dialing an extension.

Allow Callers to Break out from Voicemail

This topic describes how to allow callers to break out from voicemail, and access the operator.

Background information

For technical support, doctor office or sales manager, they do need someone available in case of any emergencies after hours. When callers access the voicemail, it would be nice to allow the callers to press `0` to get to the operator directly in emergency. Otherwise, they have to hang up and redial.

Procedure

You can specify an IVR or an extension for answering such emergency calls.

1. Log in to PBX web portal, go to **Call Features > Voicemail**.
2. In the **Caller Options** section, select the checkbox of **Allow callers to press 0 to break out from voicemail**.
3. In the **Destination** drop-down list, select a destination.
    - **IVR**: Forward the call to an IVR.
    - **Extension**: Forward the call to the specific extension.
4. Click **Save** and **Apply**.

What to do next

If a custom greeting is used for voicemail, update the greeting that would instruct callers to press 0 to break out from voicemail.

Allow Callers to Review Voicemail Messages

Callers can review their voicemail messages after recording. This is important for callers to confirm whether the message is appropriate. This topic describes how to allow callers to review voicemail messages.

Procedure

1. Log in to PBX web portal, go to **Call Features > Voicemail**.
2. In the **Caller Options** section, select the checkbox of **Allow callers to review message**.
3. Click **Save** and **Apply**.

Global Voicemail Settings

The topic describes the global voicemail message settings, including caller options, message options, and greeting options.

Caller options

Setting	Description
Allow callers to press 0 to break out from voicemail	Allow callers to press 0 to exit the voicemail, and reach a specific IVR or an extension.
Allow callers to dial extension	Allow callers to dial other extensions.
Allow callers to press 5 for leaving a message	Allow callers to press 5 to leave a voicemail message after greeting, instead of auto starting recording immediately.
Allow callers to review message	Allow callers to review his/her voicemail message after recording.

Message options

Settings	Description
Max Message Time(s)	Set the maximum duration of one voicemail message. The default maximum voicemail duration that callers can leave is 600 seconds (10 minutes).
Min Message Time(s)	Set the minimum duration of one voicemail message. The default minimum voicemail duration that callers must leave is 2 seconds.

Greeting Options

Settings	Description
Max Greeting Time(s)	Set the maximum greeting duration that is played to caller.

Settings	Description
	The default maximum greeting duration is 60 seconds (1 minute).
Global Voicemail Greeting	Select the greeting that is applied to all extensions.

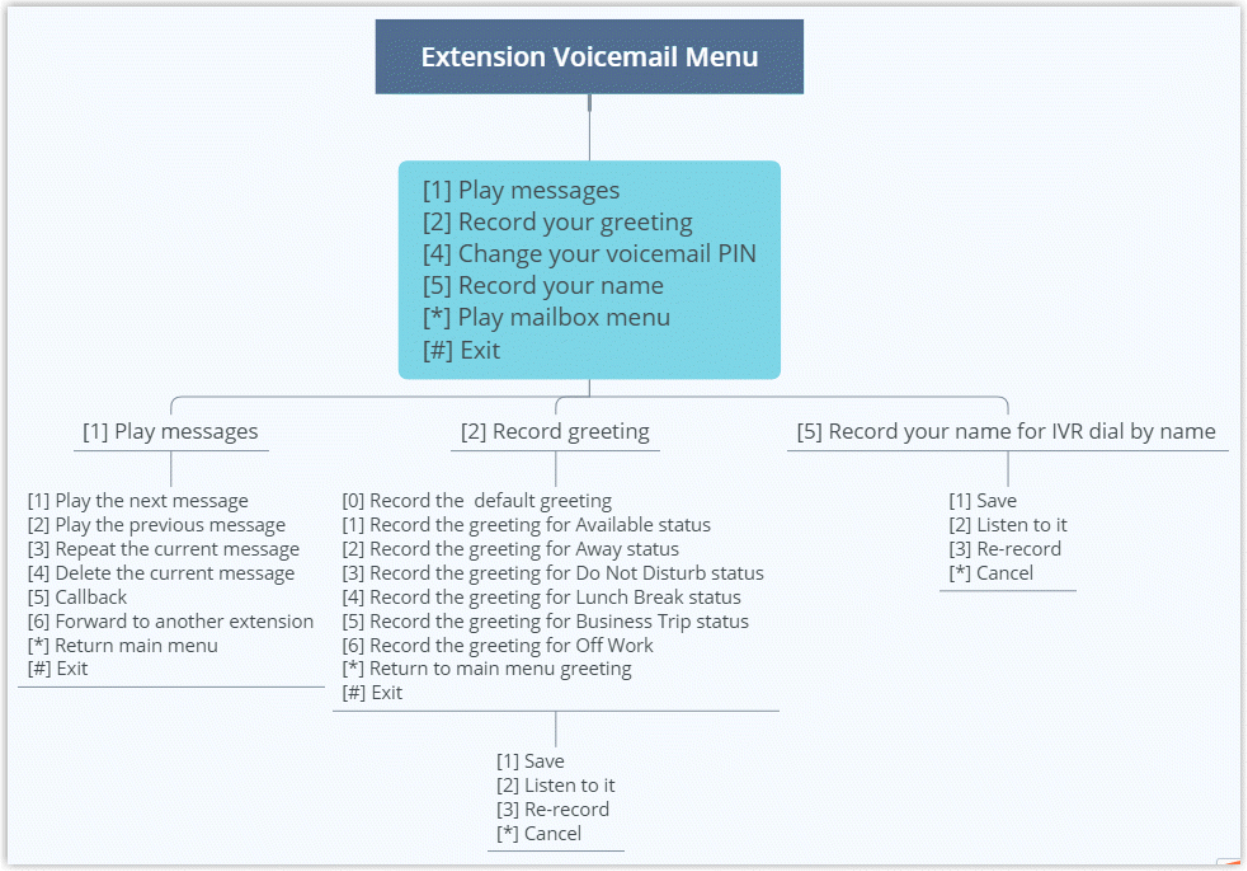
Voicemail Transcription

Decide to enable or disable Voicemail Transcription feature. For more information, see Enable or Disable Voicemail Transcription.

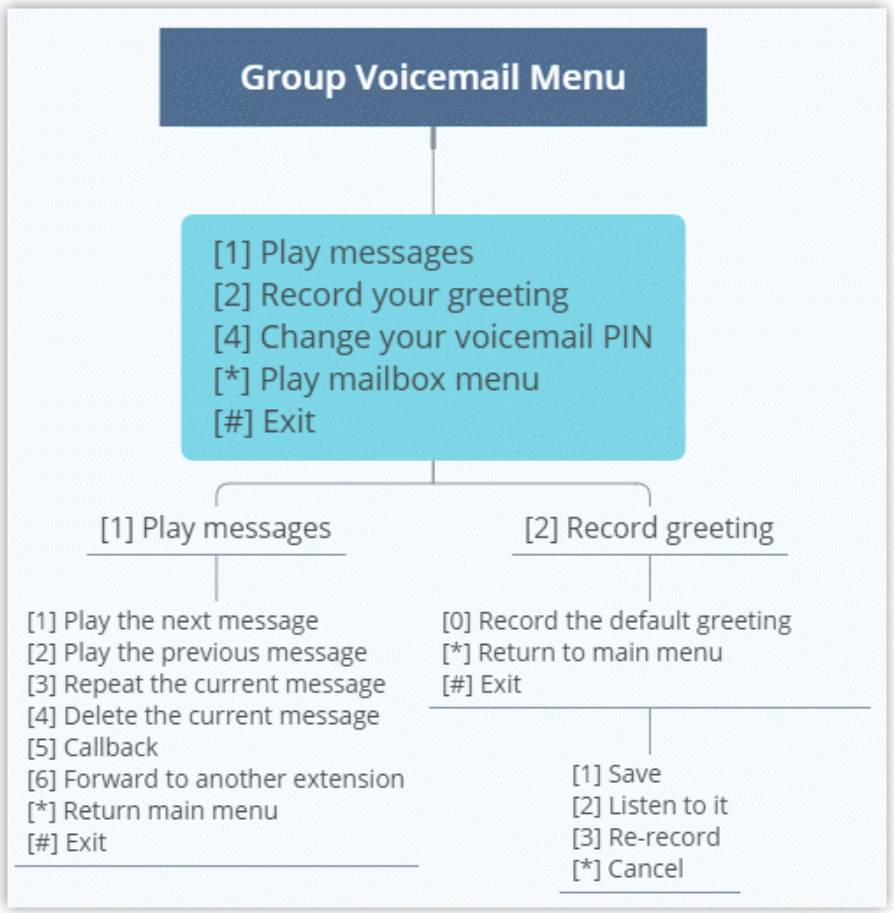
Voicemail Menu Options

This topic shows the quick reference of voicemail menu.

Extension voicemail menu



Group voicemail menu



Voicemail Capacity and Limitations

This topic describes the voicemail capacity and limitation for a voicemail.

Default capacity and limitations for each mailbox

Voicemail box capacity has a limit of 100 messages with maximum 10 minutes for each message. Once they hit that limit, the system auto deletes the old voicemail messages.

There is no limit of the time to keep the voicemail on PBX.

Adjust the capacity and limitations for each mailbox

- To adjust the capacity of voicemail box, see Auto Cleanup Voicemail Messages.
- To adjust the limitation of maximum message time, see Limit Voicemail Message Length.

IVR

Interactive Voice Response (IVR) Overview

Yeastar P-Series Cloud Edition integrates a free IVR system. This topic describes what is IVR, what you can do with IVR, and what is multi-level IVR.

What is IVR?

Interactive Voice Response (IVR) is an automated telephony technology that interacts with callers, gathers information, and routes calls to the appropriate destinations. IVR can act as a virtual receptionist to handle large volumes of calls. It means that you don't need a dedicated person to redirect calls to appropriate departments. With IVR, customers can get quick response or access appropriate service on their own.

What you can do with IVR?

Yeastar IVR uses customizable voice prompts to provide callers with instructions and directions for accessing information via phone, such as "press 1 for sales, and press 2 to leave a message.". IVR connects callers to individuals, departments, call queues, etc, based on the customers' selections from voice menus.

Multi-level IVR is an alternative that allows you to assign a new IVR to an IVR option, and provides more powerful options to route incoming calls. Multi-level IVR gives you the flexibility to classify the menu of an interaction, such as divides a sales department into regions, and routes calls more precisely.

You can customize your IVR to provide a seamless experience.

For customer

- Play personal greeting to make the customer feel welcome.
- Allow customer to leave a voicemail.
- Allow customer to call employees directly by dialing extension or by name.

For employee

- Allow employees to make an outbound call via an IVR.
- Allow employees to check voicemail via an IVR.
- Allow employees to remotely change IVR prompt by dialing the feature code #9.

IVR keypress events

There are three types of keypress events:

- **Menu options**: The number keys, # key and * key for users to access a desired destination.
- **Timeout**: If no input is detected after the configured timeout, the PBX will forward the call based on the configuration.
- **Invalid**: When an invalid key is pressed, route the call to a desired destination.

Keypress destination

The following options are available for you to assign to the keypress events:

- **Hang Up**: End the current call.
- **Extension**: Route the call to the specified extension.
- **Extension Voicemail**: Allow callers to leave a message for the specified extension.
- **Group Voicemail**: Allow callers to leave a message for a queue, a ring group, or a custom group.
- **IVR**: Allow callers to enter another IVR menu.
- **Ring Group**: Route the call to a specified ring group.
- **Queue**: Route the call to a specified queue.
- **Conference**: Route the call to a specified conference.
- **Dial by Name**: Allow callers to place a call by extension user's name.

  For more information, see Allow Callers to Dial by Name via IVR.
- **External Number**: Route the call to an external number.
- **Play Prompt and Exit**: Play a custom prompt, and then hang up the call.
- **Play Prompt and Return to IVR**: Play a custom prompt, and then back to the IVR.
- **Play IVR Prompt**: Play the IVR prompt, and then hang up the call.

> **Note:**
> - The option is only available for **Invalid Input Destination**.
> - When callers enter a DTMF digit that is not defined in the IVR, the system would repeat the IVR prompt. If the play counts of the IVR prompt reach the maximum number of times, the system would directly hang up the call.

Set up an IVR

Yeastar P-Series Cloud Edition provides easy-to-create menus that allow you to set up an IVR and keep up with changing requirements. This topic describes how to set up an IVR.

Prerequisites

Before you set up an IVR, record a custom prompt or upload a custom prompt to provide callers with the IVR menu.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, click **Add**.
2. In the **Basic** tab, set the basic settings of IVR.
   - **Number**: Specify a virtual number for callers to access the IVR.

     The default IVR number range is from 6200 to 6299.
   - **Name**: Enter an IVR name to help you identify it.
   - **Prompt**: Set the IVR prompt that plays greeting and explains the IVR menu options to callers.

     The default prompt is "Dial the extension number or press 0 for operator".

     You can select up to 5 audio files, and the system plays the audio files in order.
   - **Prompt Repeat Count**: Set how many times to play the prompt when the caller remains inactive during the **Response Timeout(s)**.
   - **Response Timeout(s)**: Set how long (in seconds) to wait for the caller to operate.
   - **Digit Timeout(s)**: Set how long (in seconds) to wait for the caller to enter the next digit.
   - **IVR Alert Info**: Optional. Set an "alert info text" to add to Alert-info header in INVITE request for IVR calls.

     When receiving an IVR call, the phone will inspect "Alert-Info" header to determine which ring tone it should use for ringing.
   - **Dial Extensions**: Whether to allow callers to dial specific extension numbers via IVR.
     - **Disable**: Disable to dial extensions via IVR.
     - **All Extensions**: Allow the callers to dial all the extension numbers.
     - **Allowed Extensions**: Select the extensions that the callers can dial.
     - **Restricted Extensions**: Select the extensions that the callers can NOT dial.

- **Dial Outbound Routes**: Whether to allow callers to make outbound calls via IVR.
- **Dial to Check Voicemail**: Whether to allow users to check voicemail via IVR.
- **Dial #9 to Modify IVR Prompt**: Whether to allow users to dial the feature code #9 to record and apply a new IVR prompt.

> **Note:**
> If the IVR prompt is replaced successfully, the previous voice prompt will be removed from the IVR prompt setting, and the new voice prompt will be retained.

3. Click the **Key Press Event** tab, set up an IVR menu.
   - To set key events to route calls to different destinations based on time, see Set Key Events Based on Time Conditions.
   - To set key events to always route calls to the designated destination, do as follows:
     a. In the **Key Press** drop-down list, select a key event for each key: 0-9, *, and #.
     b. In the **Response Timeout** drop-down list, select a call routing destination if the caller remains inactive within the **Prompt Repeat Count**.
     c. In the **Invalid Input Destination** drop-down list, select a call routing destination if the caller enters a digit that is not defined in the IVR.
     d. **Optional:** Select the checkbox of **Allow Opt-out of Call Recording**.

        When the call is routed to the key press destination, the call would not be recorded even **Call Recording** is enabled.
4. Click **Save** and **Apply**.

What to do next

Set up an inbound route, and specify the destination to the IVR.

Related information

Allow Callers to Dial Extensions via IVR
Allow Callers to Dial by Name via IVR
Allow Callers to Dial Outbound Calls via IVR
Allow Callers to Change IVR Prompt Remotely
Forward Incoming Calls to an External Number via IVR
Set up a Multi-language IVR
Set Key Events Based on Time Conditions

Set up IVR Prompts

A custom greeting and prompt allow you to provide a more personalized experience for your customers. This topic describes how to set up IVR prompts according to your IVR menu.

IVR prompt types

Generally, an IVR prompt consists of several pieces of information:

- **Welcome greeting**: Welcome greeting is the first message that callers hear when they call in an IVR.

  For example, "Thank you for calling Yeastar".
- **Menu prompt**: Present callers with a series of options.

  For example, "If you got something urgent, please press 1 to contact our support. To leave a voicemail, please press 2".
- **Goodbye greeting**: Play the greeting before ending a call.

Prepare audio files for IVR prompt

The PBX system has a default IVR prompt. You can customize IVR prompt using a single audio file or multiple audio clips.

Customize IVR prompt by a single audio file

You can record greeting, IVR menu, or any messages in a single audio file. It is easy to manage and reduce the number of prompts.

Customize IVR prompt by multiple audio clips

Yeastar IVR also allows you to specify up to 5 different audio files as IVR prompt. The system plays the audio files in order when a customer calls in IVR.

It is better to divide your IVR prompt into multiple audio clips in the following scenarios:

- Modify the IVR prompt frequently.

  Every time you modify the IVR menu, you need to update IVR prompt. Divide your IVR prompt into multiple audio clips according to the content, such as clip 1 for Welcome greeting, clip 2 for menu prompt, and so on. Next time, when you need to change the IVR prompt, just replace the specific clip.
- A single audio file exceeds the limit.

The uploaded file should meet the audio file requirements. You can not upload an audio file larger than 8 MB. Divide the audio file into multiple audio clips to solve this issue.

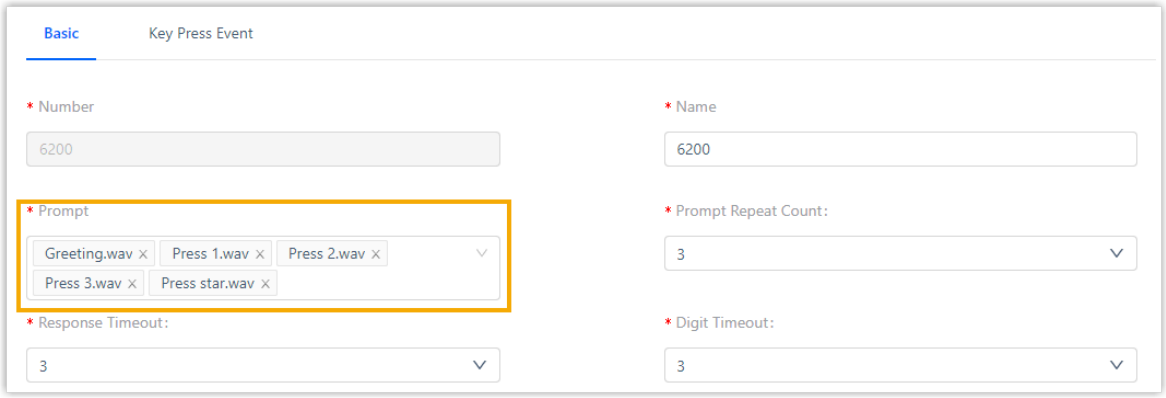
Update the IVR prompts

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**, upload a custom prompt or record a custom prompt.

   > **Note:**
   > The uploaded file should meet the audio file requirements.

2. Go to **Call Features > IVR**, edit the desired IVR.
3. In the **Prompt** drop-down list, select your custom prompts.

   You can select up to 5 audio files, and the system plays the audio files in order.

4. In the **Prompt Repeat Count** drop-down list, select prompt repeat times.
5. Click **Save** and **Apply**.

Related information

Allow Callers to Change IVR Prompt Remotely

Allow Callers to Dial Extensions via IVR

This topic describes how to allow callers to dial extensions directly via an IVR.

Background information

For new customers, IVR can help them reach the desired employee or department. But for old customers, it is inconvenient for them to listen to audio prompts and make selections to reach the right employee or department, even they know the extension number.

For the callers who know the extension number, it is better to allow them to dial an extension number directly.

Prerequisites

Before you set up dialing extension directly via an IVR, update your IVR prompt that would instruct callers to dial an extension number.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. In the **Prompt** drop-down list, select the updated IVR prompt.
3. In the **Dial Extensions** drop-down list, select which extension as available or available for callers to dial.
    - **All Extensions**: The callers can dial all the extensions.
    - **Allowed Extensions**: The callers can dial the selected extensions.
    - **Restricted Extensions**: The callers can dial any extensions except the restricted extensions.
4. Click **Save** and **Apply**.

Allow Callers to Dial by Name via IVR

For the customers who don't remember an employee's extension number, you can allow them to reach the employee by entering the first three letters of the employee's first name or last name in an IVR. It is easier for customers to get to the right person.

Restrictions

The **Dial by Name** feature supports to search the extension users whose caller ID names are composed of English letters or Mandarin phonetic symbols.

Procedure

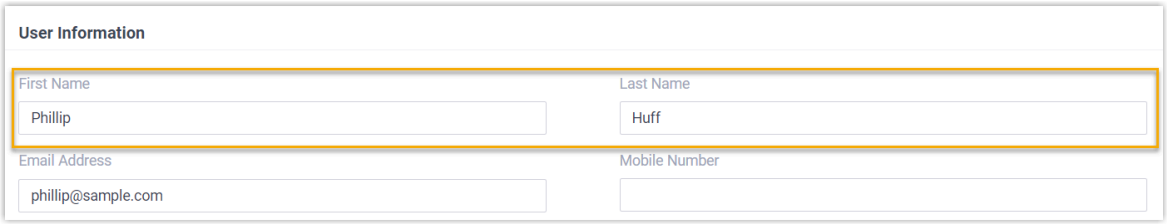
1. Specify an extension's caller ID name
2. Set display format for extensions' caller ID name

3. Customize IVR prompt
4. Configure an IVR

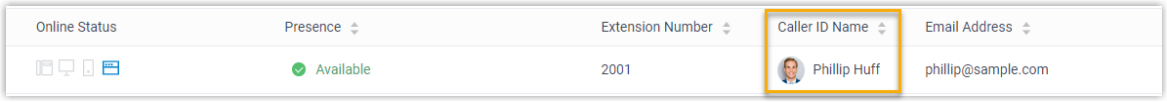
Specify an extension's caller ID name

When using **Dial by Name**, the IVR performs a search on an extension's caller ID name, which is composed of the first name and last name of the extension user. Therefore, make sure you have configured the caller ID name for extensions.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. In the **User Information** section, specify the **First Name** and **Last Name** for the extension.



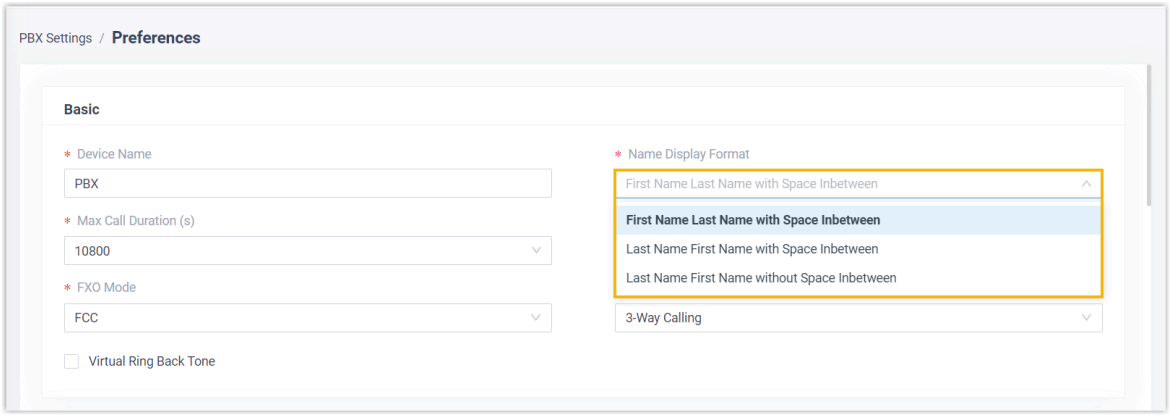
3. Click **Save** and **Apply**.



Set display format for extensions' caller ID name

When searching for extensions by name, the IVR starts from the beginning of extensions' caller ID name. By setting display format for extensions' caller ID name, you can determine how callers can reach employee - either by **First Name** or **Last Name**.

1. Log in to PBX web portal, go to **PBX Settings > Preferences**.
2. In the **Name Display Format** drop-down list, select a display format.



- To allow callers to reach extension users by first name, select **First Name Last Name with Space Inbetween**.
- To allow callers to reach extension users by last name, select **Last Name First Name with Space Inbetween** or **Last Name First Name without Space In-between**.

3. Click **Save** and **Apply**.

Customize IVR prompt

You need to prepare a custom IVR prompt, instructing the callers to press a specific key in an IVR to enter the **Dial by Name** feature.

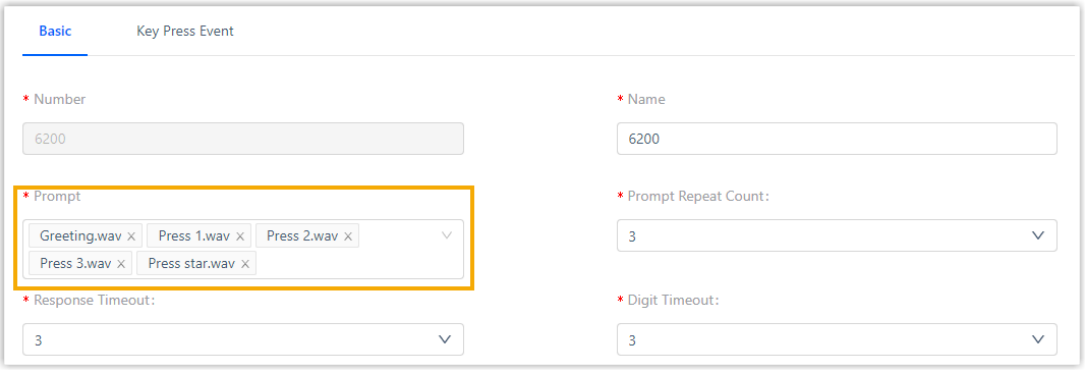
1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**.
2. Upload a custom prompt or record a custom prompt.

> **Note:**
> The uploaded file should meet the audio file requirements.

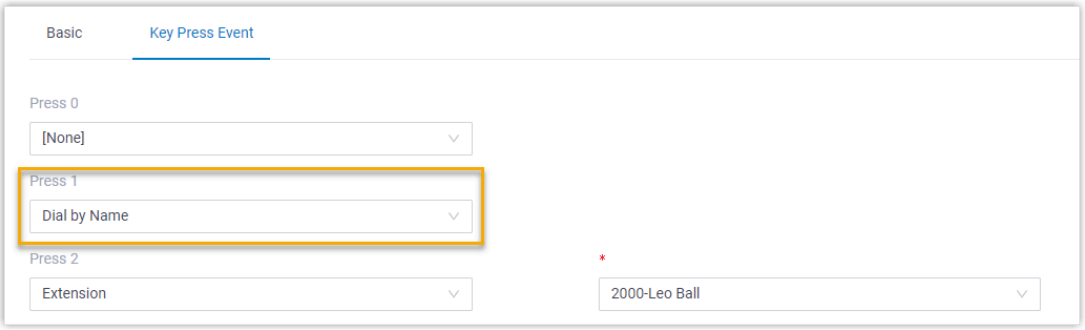
Configure an IVR

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. Update the IVR prompt.
   a. Go to the **Basic** tab.
   b. In the **Prompt** drop-down list, select your custom prompts.

   You can select up to 5 audio files, and the system plays the audio files in order.



3. Set up the IVR keypress destination.
    a. Go to the **Key Press Event** tab.
    b. In the drop-down list of a key press, select **Dial by Name**.



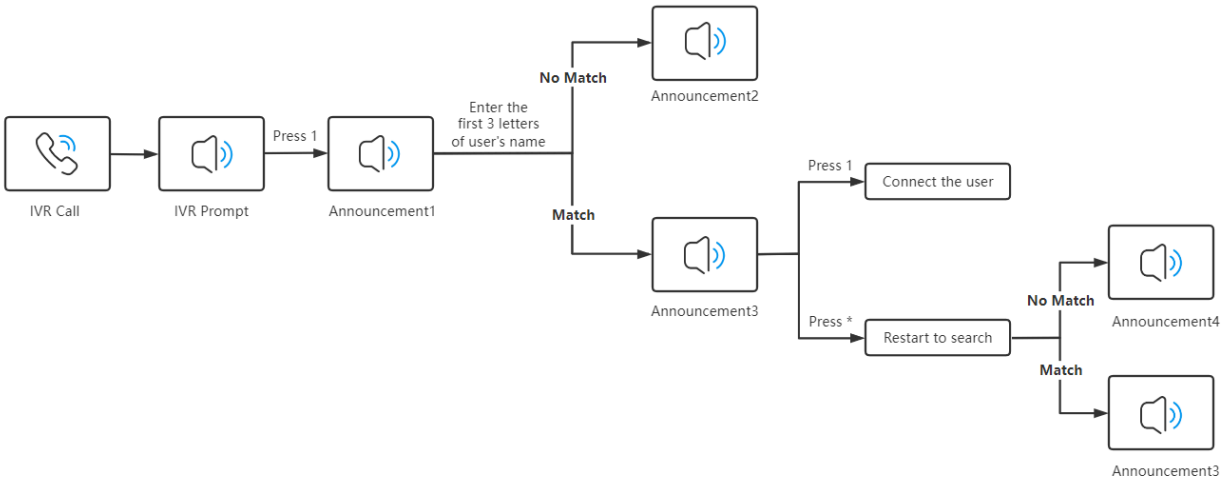
4. Click **Save** and **Apply**.

Result

Customers can call to the IVR, and quickly search the desired extension user by entering the first 3 letters of his/her name.

Example

We provide an example to help you understand the workflow of dial-by-name. In this example, the display format for extensions' caller ID name is set to **First Name Last Name with Space Inbetween** and the destination of key "1" is set to **Dial by Name**.



1. A caller calls into an IVR and hears the IVR prompt.
2. The caller presses "1" to enter the **Dial by Name** feature.
3. After hearing the announcement1, the caller enters the first 3 letters of an extension user's first name.

   For example, to search an extension user with the name "Phillips Huff", the caller needs to dial 7-4-4 (indicating `P H I`) on the phone's keypad.
4. The IVR will look for the best match and play the corresponding announcement.
   - If there is no matched user, the IVR plays announcement2.
   - If there is a matched user, the IVR plays announcement3, providing the extension user's name and extension number, as well as the following operation instructions:
     - **1**: If this is the person you are looking for, press 1 now.

       The IVR will send the call to the extension user.
     - *: Otherwise please press star now.

       The IVR will start a new search, and provide another matched user's information to the caller. If there is no more matched user, IVR will play announcement4.

Default announcement of Dial by Name

Yeastar provides the default announcements when the caller selects the **Dial by Name** option. An announcement is played in the following scenarios:

Announcement	scenario
Welcome to the directory. Please enter the first three letters of your party's first name,	Play when the caller presses a key to dial by name.

Announcement	scenario
using your touch tone keypad, use the 7 key for Q, and the 9 key for Z.	
No directory entries match your search.	Play when there is no matching directory entries after the caller enters three letters.
[Name] extension [Number] If this is the person you are looking for, press 1 now, otherwise please press star now.	Play when there are matching directory entries after the caller enters three letters.
There are no more compatible entries in the directory.	Play when there are no more compatible entries in the directory after the caller presses * key to search.

Allow Callers to Dial Outbound Calls via IVR

This topic describes how to allow callers to dial outbound calls in an IVR.

Background information

Dialing outbound calls via an IVR is useful when you interconnect two PBXs between headquarters and branch, and only set an IVR on headquarters PBX. You can allow the customers to dial the headquarters' extension number to contact the employees or departments in branch directly.

Prerequisites

- Set up the appropriate outbound route and inbound route on the two interconnected PBXs.
- Upload or record IVR prompt that would instruct customers to dial an outbound call.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. In the **Prompt** drop-down list, select the updated IVR prompt.
3. Select the checkbox of **Dial Outbound Routes**.
4. Select the desired outbound route from the **Available** box to the **Selected** box.
5. Click **Save** and **Apply**.

Allow Callers to Change IVR Prompt Remotely

In case of emergency (e.g. the office needs to close early due to bad weather), you may need to change IVR prompt. Instead of logging in to PBX with a computer to change IVR prompt, you can just make a call to the IVR, then dial a specific feature code to record and apply a new IVR prompt.

Restrictions

- The number of custom prompts does NOT reach the maximum limit. Otherwise, users can NOT record new voice prompt for the IVR.
- A maximum of 2-minute recording time is allowed.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. In the **Basic** tab, select the checkbox of **Dial #9 to Modify IVR Prompt**.
3. In the **IVR Prompt Modify Password** field, enter a password for authentication.

   Callers need to enter the password to authenticate, so as to change the IVR prompt.
4. Click **Save** and **Apply**.

Result

Users can call to an IVR, dial #9 and enter the password, then follow the voice prompt to record a new IVR prompt on their phones.

If IVR prompt is replaced successfully, the previous voice prompt will be removed from the IVR setting, and the new voice prompt will be retained.

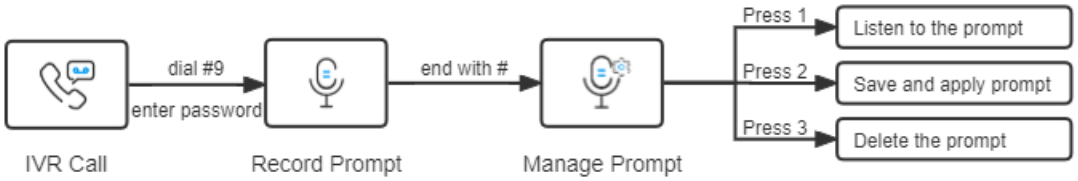
Note:

The new voice prompt is save on **PBX Settings > Voice Prompt > Custom Prompt**, with a prompt name in the format of **IVR*{ivr_number}*Date*{date}*Number*{extension_number}***.



Example

We provide an example to help you understand the workflow of remotely changing IVR prompt.



1. In an IVR call, a caller dials #9, then enter the password to authenticate.
2. After hearing the beep tone, the caller starts recording the prompt. When done, press # key.
3. The caller can press a specific key to manage the prompt:
    - **1**: Listen to the prompt.
    - **2**: Save and apply the prompt to the IVR.
    - **3**: Delete the prompt.

Forward Incoming Calls to an External Number via IVR

This topic describes how to allow callers to reach a specific external number in an IVR.

Background information

Forward Incoming Calls to an External Number with IVR is typical and important for 24x7 services, such as Doctor Answering Services and IT Support Services.

For Doctor Answering Services

When a patient calls in an hospital IVR, the patient can press a key to reach the external Doctor Answering Service to schedule an appointment or ask health questions and medical questions.

For IT Support Services

When your customers call in your office IVR after hours, you can give them an option to connect to an emergency support line. This emergency support line can be a Maintenance Engineer's mobile phone number.

Prerequisites

Before you allow callers to reach a specific external number in an IVR, update your IVR prompt that would instruct callers to press a key to reach the external number.

Procedure

1. Log in to PBX web portal, go to **Call Features > IVR**, edit the desired IVR.
2. In the **Prompt** drop-down list, select the updated IVR prompt.
3. Click **Key Press Event** tab.
4. Select a key to set key press event to **External Number**.
5. **Optional:** In the **Prefix** field, enter the prefix of outbound route so that PBX can successfully route incoming calls to external number.
    - If the **Strip** of outbound route is not set, you don't have to set the **Prefix**.
    - If the **Strip** of outbound route is set, you need to set the **Prefix** according to the **Patterns** of outbound route.
6. Enter the external number, such as a Doctor Answering Service number or a mobile phone number.
7. Click **Save** and **Apply**.

IVR Configuration Example

This topic shows the examples of single IVR configuration and multi-level IVR configuration.

A Single IVR Configuration

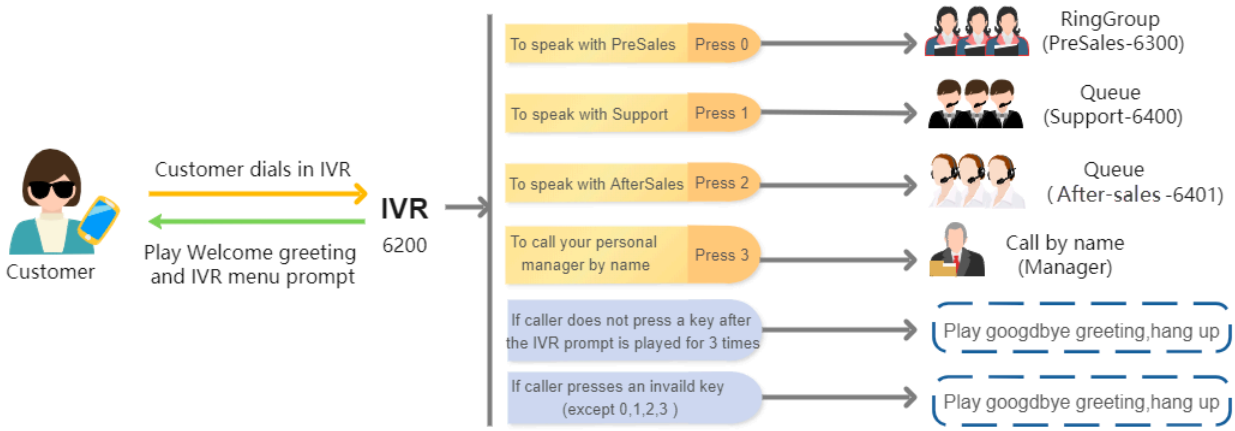
Background information

A company needs an IVR to redirect calls to Pre-sales, Support, After-sales, and personal manager.

We assume that all ring groups, call queues, audio prompts, and inbound routes used in this example are previously configured.

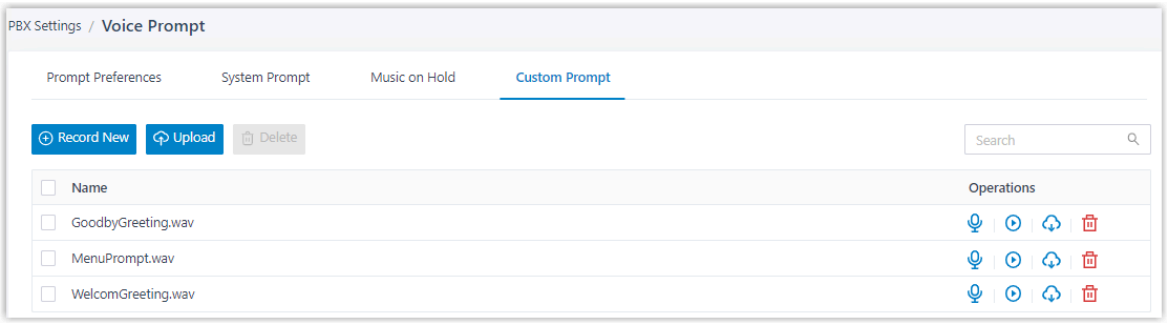
Step1. Design an IVR

When the customers dial in IVR (6200), they can access different service based on their business.



Step2. Upload IVR Prompts

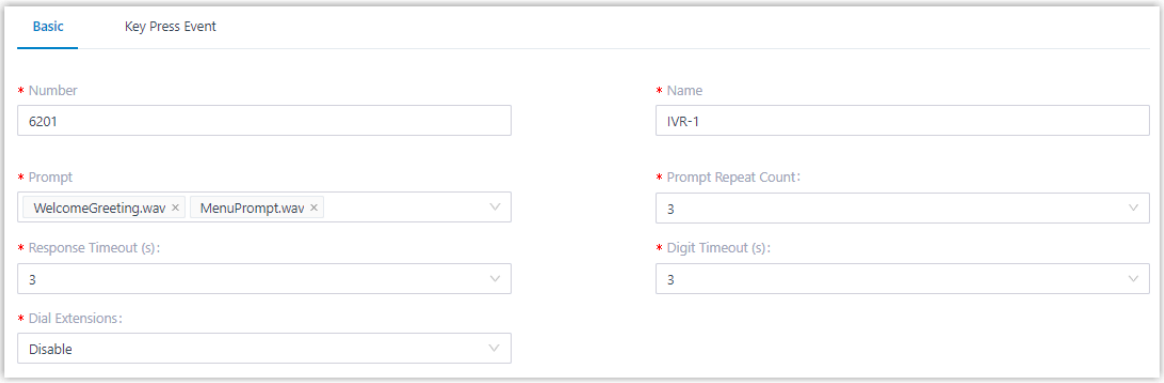
1. Go to **PBX Settings > Voice Prompt > Custom Prompts**, click **Upload**.
2. Select the audio files to upload.



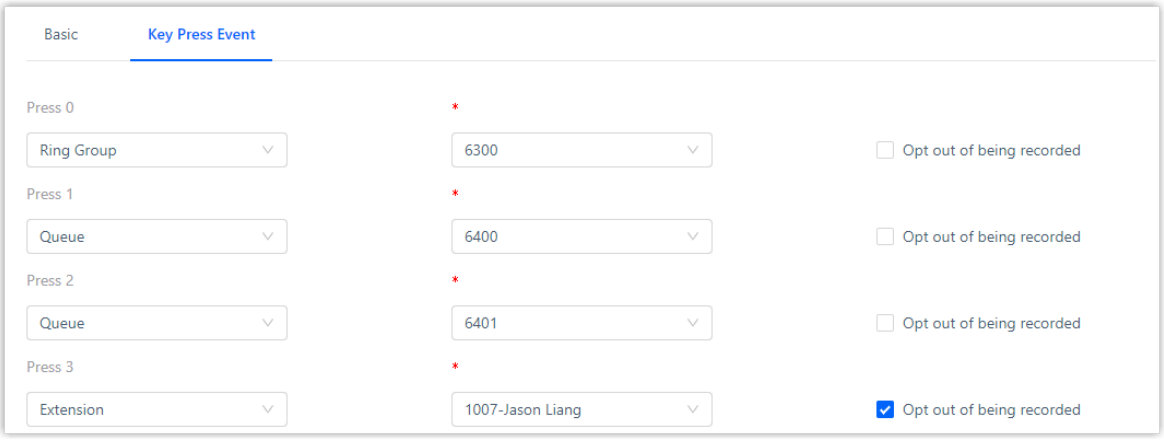
3. Click **Save** and **Apply**.

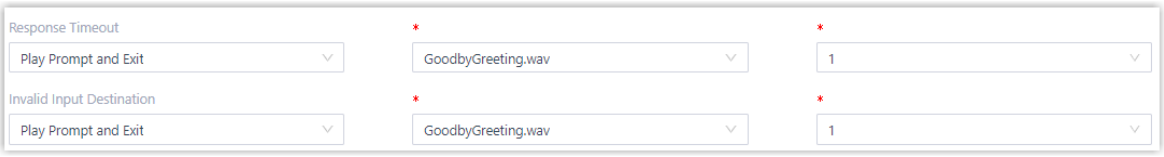
Step3. Set up an IVR

1. Go to **Call Features > IVR**, click **Add**.
2. In the **Basic** tab, set the basic settings of IVR.



3. In the **Key Press Event** tab, set up an IVR menu.





4. Click **Save** and **Apply**.

Multi-level IVR Configuration

Background information

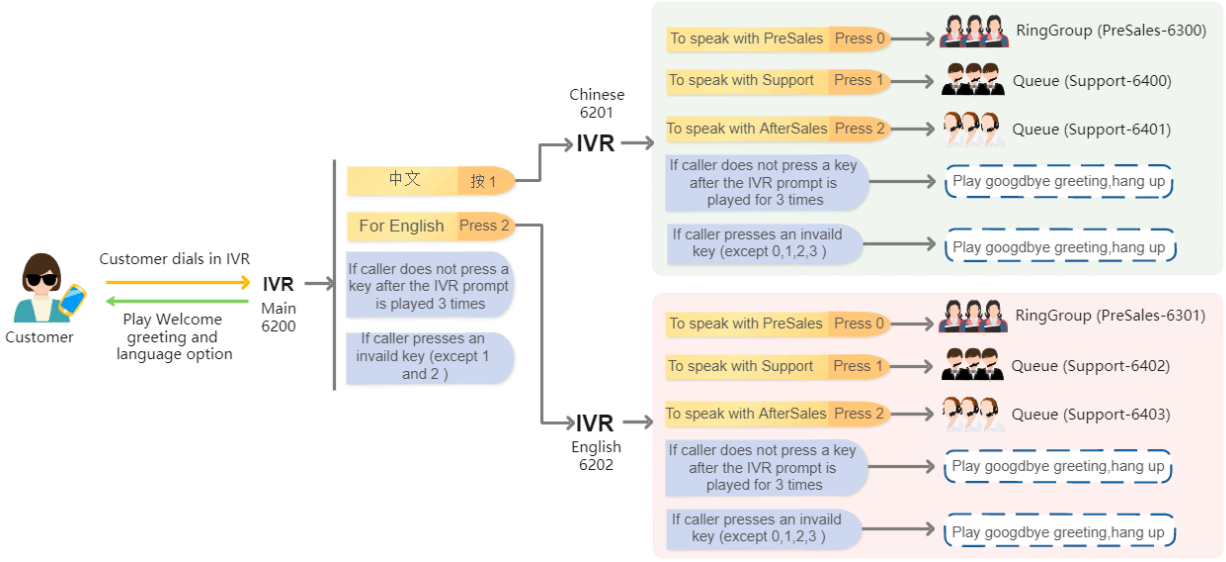
As business expands, the company needs to offer callers a bilingual auto-attendant feature based on their language selection. When the customer dials in to IVR, they can select a specific language to be used when playing prompts.

To achieve this, company needs to upgrade its IVR system allowing support of multi-language. We assume that all ring groups, call queues, extensions, audio prompts, and inbound routes used in this example are previously configured.

Step1. Design IVRs

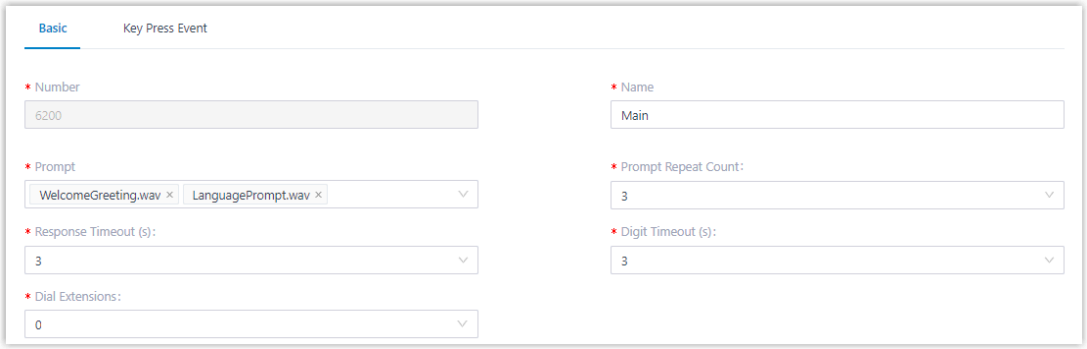
When the customers dial in to IVR-Main (6200), they can select a specific language to use.

- If customers select Chinese, the call will be redirected to IVR-Chinese (6201).
- If customers select English, the call will be redirected to IVR-English (6202).



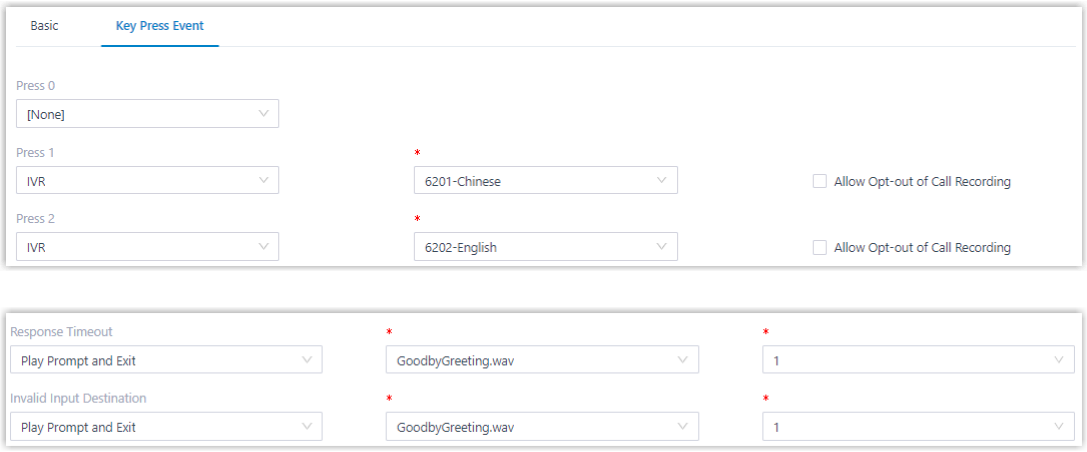
Step2. Set up IVRs

1. Set up the different IVRs with the same configuration for different language as shown in a single IVR configuration.
   - IVR-1 (6201)
   - IVR-2 (6202)
2. Set up main IVR-Main (6200).
   a. In the **Basic** tab, set the basic settings of IVR.



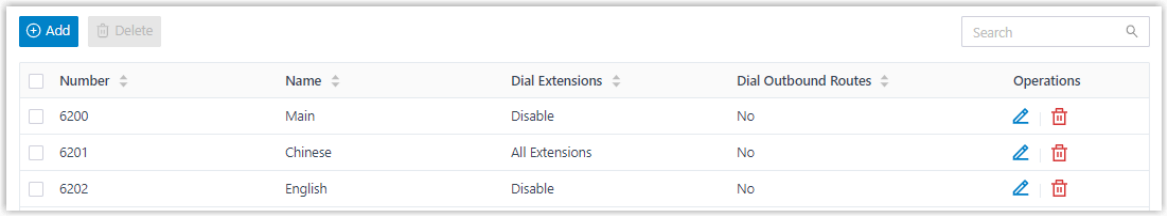
   b. In the **Key Press Event** tab, set up an IVR menu.

- Specify IVR-Chinese (6201) for key 1.
- Specify IVR-English (6202) for key 2.



3. Click **Save** and **Apply**.

   The following figure displays the different IVRs created.



Advanced IVR Settings

Set up a Multi-language IVR

A multi-language IVR allows callers to select their language preferences and provides them with system prompts in their preferred language. You can set the IVR to route the callers to designated destinations based on different language selections, enabling better handling of customer calls and improved customer satisfaction.

Requirements

Yeastar P-Series Cloud Edition is 84.14.0.24 or later.

Step 1. Design a multi-language IVR

Design a multi-language IVR and outlines the following information:

- Language options the IVR supports

- Key for each language option
- Call routing destination for each language option

> **Note:**
> Make sure that you have created the respective destinations on PBX.

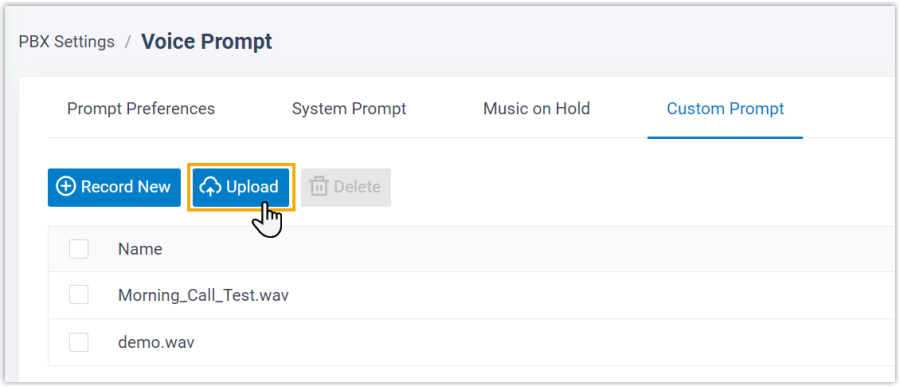
For example, design an IVR with 2 language options (English and Mandarin) supported.

Language Option	Key	Destination
English	key 1	IVR: English (6202)
Mandarin	key 2	Queue: Mandarin-agents (6400)

Step 2. Upload the IVR prompt

1. Prepare a voice prompt for the IVR to announce the available language options and their respective key actions.

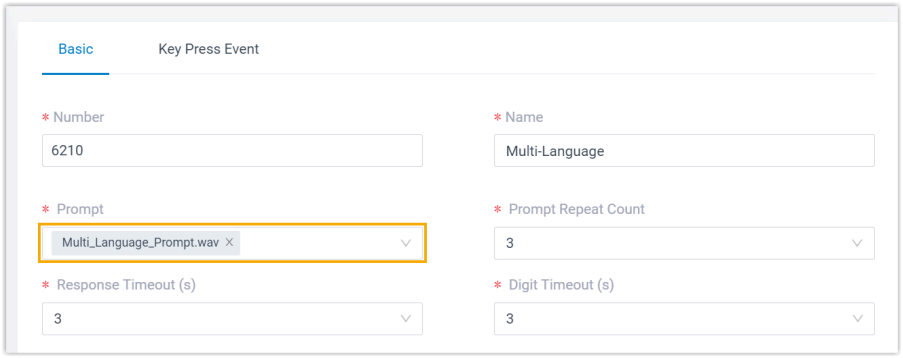
   For example, "Welcome. For English, please press 1. For Mandarin, please press 2".
2. Upload the voice prompt to the PBX system.
   a. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompts**.
   b. Click **Upload**, and select the audio file to upload.



   c. Click **Save** and **Apply**.

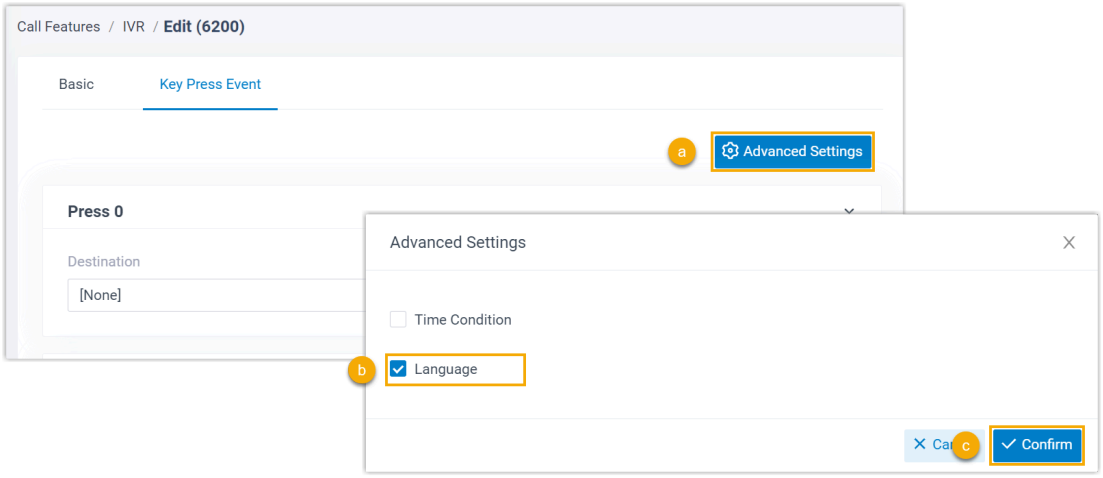
Step 3. Set up the multi-language IVR

1. On PBX web portal, go to **Call Features > IVR**, click **Add**.
2. In the **Basic** tab, configure the basic settings for the IVR.



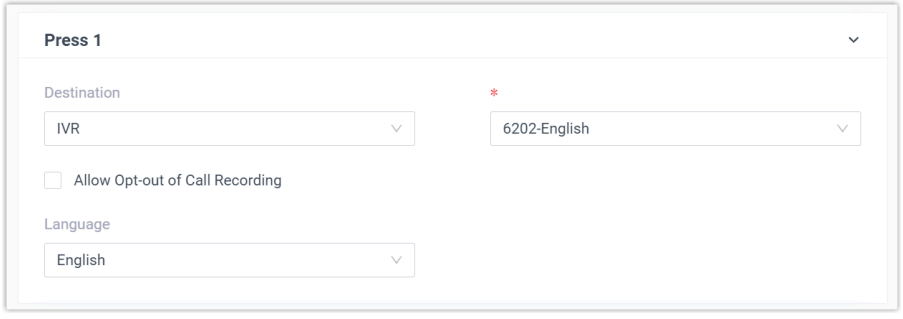
a. In the **Prompt** drop-down list, select the uploaded voice prompt.
b. Configure other settings as needed.

3. In the **Key Press Event** tab, enable multi-language setting for the IVR.



a. On the top-right corner, click **Advanced Settings**.
b. In the pop-up window, select the checkbox of **Language**.
c. Click **Confirm**.

4. Configure the corresponding key events based on your IVR design.



- **Destination**: Specify the call routing destination.

When callers press the key, the call will be routed to the specified destination.

- **Optional:** Select the checkbox of **Allow Opt-out of Call Recording**.

  If enabled, when the call is routed to the key press destination, the call would not be recorded even if **Call Recording** is enabled.
- **Language**: Select the system prompt language.

  > **Note:**
  > The displayed language options are synchronized from System Prompts (Path: **PBX Settings > Voice Prompt > System Prompt**).

  When callers press the key, all the subsequent system prompts will be played in the specified language.

5. Click **Save** and **Apply**.

Result

When calling into the IVR, callers hear the voice prompt for language preference selection. After they press the corresponding key, all subsequent system prompts (except the custom ones) will be played in the selected language, and the call will be routed to the respective key press destination.

Set Key Events Based on Time Conditions

Yeastar P-Series Cloud Edition supports setting IVR key press events based on time conditions, enabling the routing of incoming calls to different destinations based on the time they are received. You can apply a single time condition for all key press events or customize time conditions for each key press event.

Requirements

Yeastar P-Series Cloud Edition is 84.14.0.24 or later.

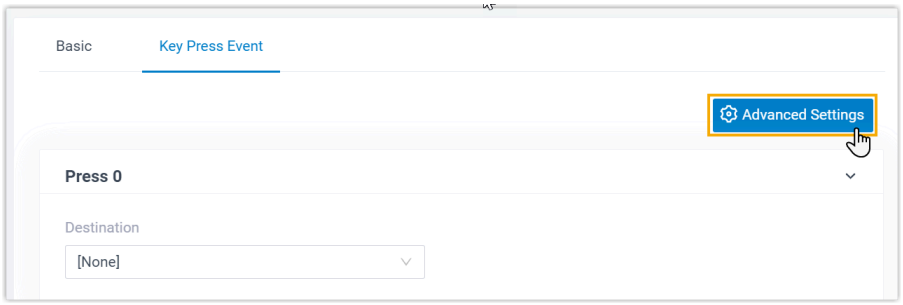
Apply the same time condition for all key press events

1. Log in to PBX web portal, go to **Call Features > IVR**.
2. Click beside the desired IVR.
3. In the **Key Press Event** tab, configure key press events to route calls according to your needs.

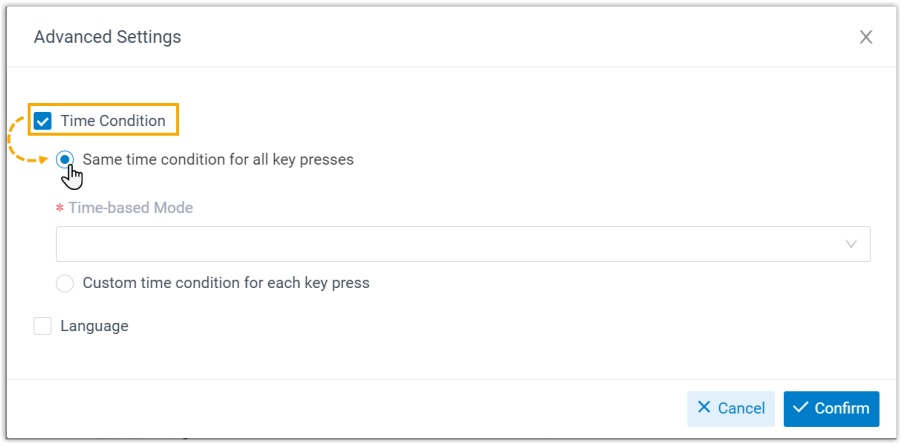
Scenario	Description
Route calls based on global business hours	All key press events route calls to specified destinations based on the global business hours, non-working hours, and holidays (Defined on **Call Control > Business Hours and Holidays**). For detailed instructions, see Route calls based on global business hours.
Route calls based on custom business hours	Create a custom business hours, and all key press events route calls to specified destinations based on the custom business hours, non-working hours, and holidays. For detailed instructions, see Route calls based on custom business hours.
Route calls based on custom time periods	Create a custom time period for each key press event, and route calls based on the custom time period accordingly. For detailed instructions, see Route calls based on custom time periods.

Route calls based on global business hours

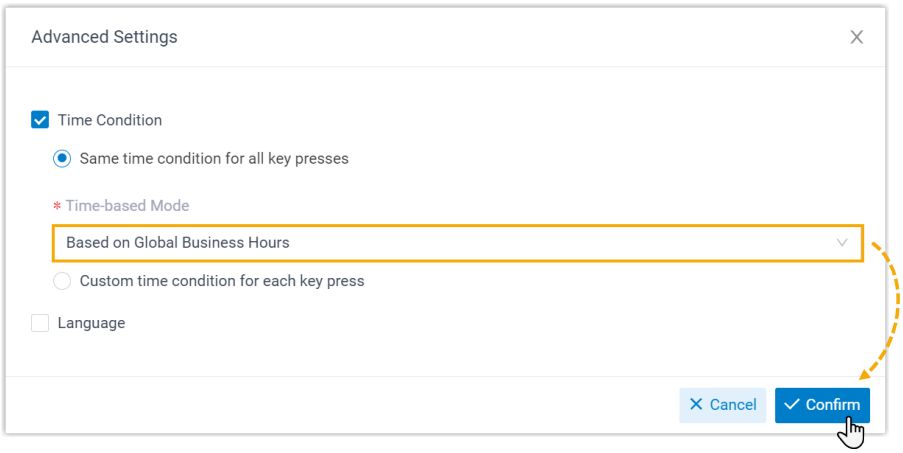
a. On the top-right corner, click **Advanced Settings**.



b. In the pop-up window, select the checkbox of **Time Condition**, and select **Same time condition for all key presses**.

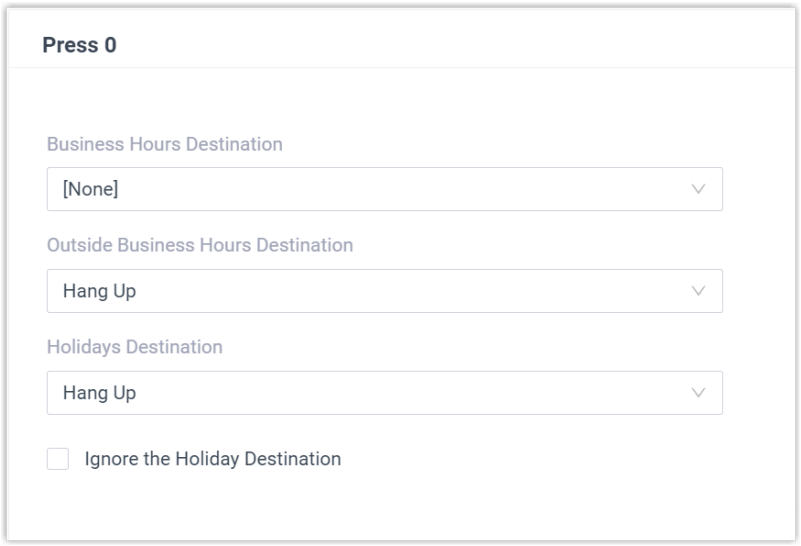


c. In the **Time-based Mode** drop-down list, select **Based on Global Business Hours**, and click **Confirm**.



The global business hours is applied to all key press events in the IVR.

d. Specify call routing destinations for a key press event based on the global business hours.



- **Business Hours Destination**
- **Outside Business Hours Destination**
- **Holidays Destination**

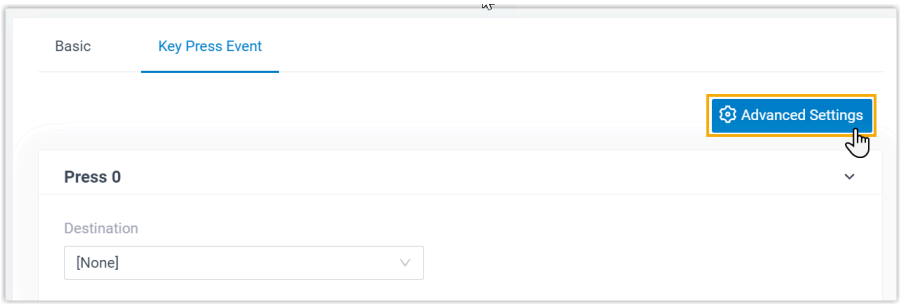
> **Note:**
> If you want to route calls based on global business hours and non-working hours during holidays, select the checkbox of **Ignore the Holiday Destination**.

e. Repeat step **d** for all the desired keys.

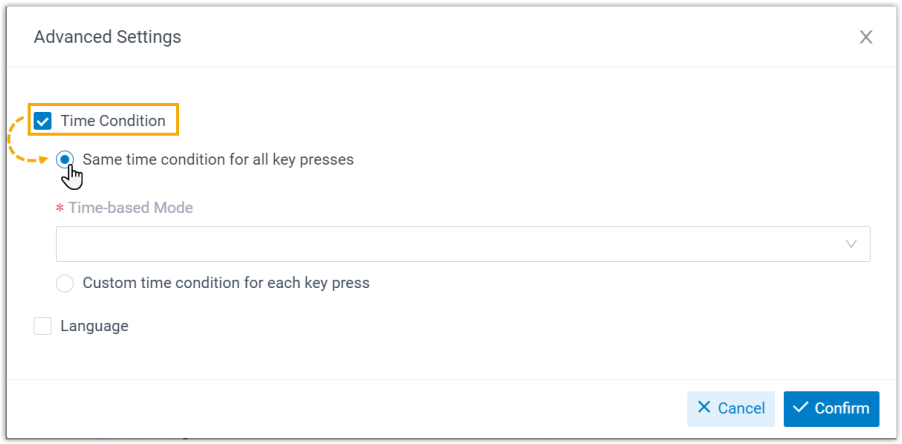
f. Click **Save** and **Apply**.

Route calls based on custom business hours

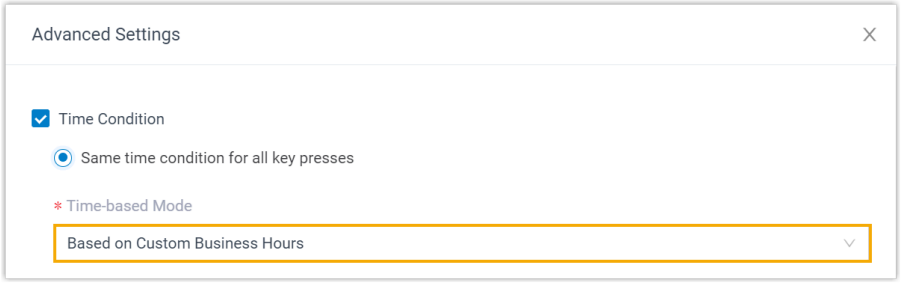
a. On the top-right corner, click **Advanced Settings**.



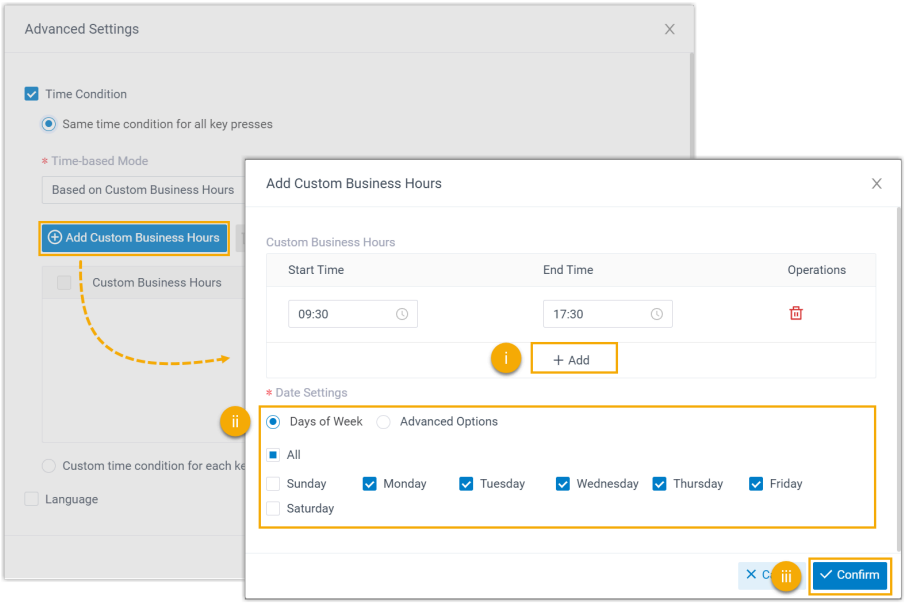
b. In the pop-up window, select the checkbox of **Time Condition**, and select **Same time condition for all key presses**.



c. In the **Time-based Mode** drop-down list, select **Based on Custom Business Hours**.



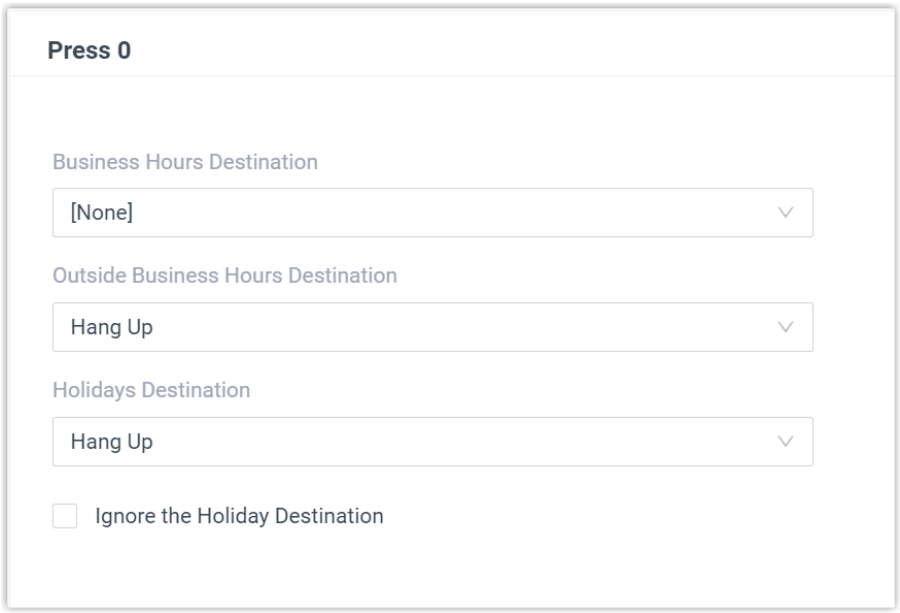
d. Click **Add Custom Business Hours** to create custom business hours.



i. Click **Add** and specify the start time and end time.

ii. In the **Date Settings** section, specify the workdays.
iii. Click **Confirm**.

e. Click **Confirm** to apply this custom business hours to all key press events in the IVR.
f. Specify call routing destinations for a key press event based on the custom business hours.



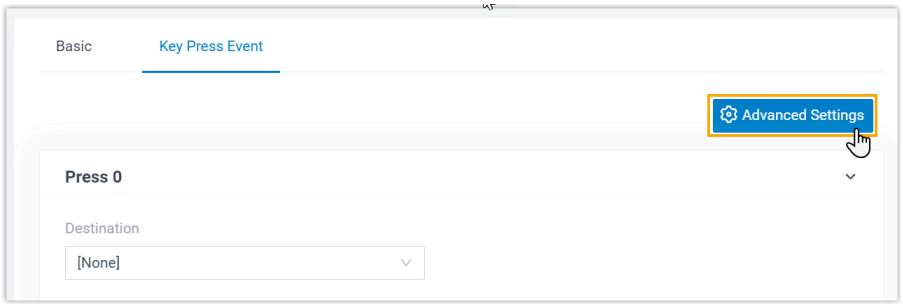
- **Business Hours Destination**
- **Outside Business Hours Destination**
- **Holidays Destination**

> **Note:**
> If you want to route incoming calls based on custom business hours and non-working hours during holidays, select the checkbox of **Ignore the Holiday Destination**.

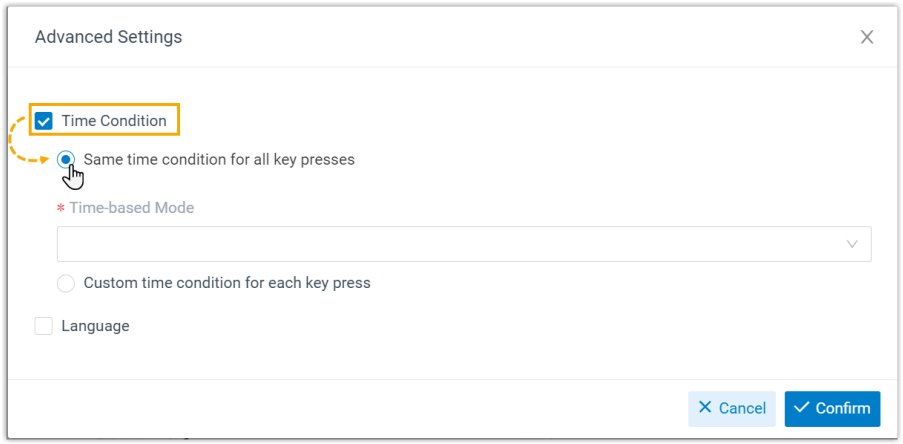
g. Repeat step **f** for all the desired keys.
h. Click **Save** and **Apply**.

Route calls based on custom time periods

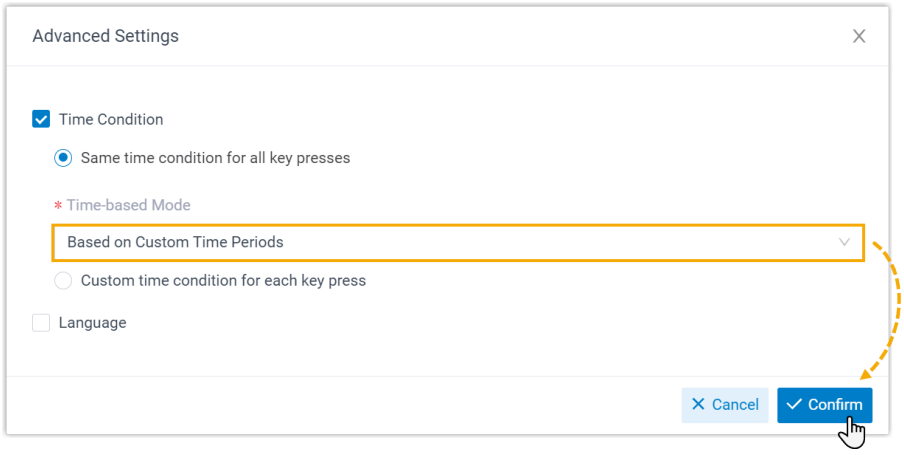
a. On the top-right corner, click **Advanced Settings**.



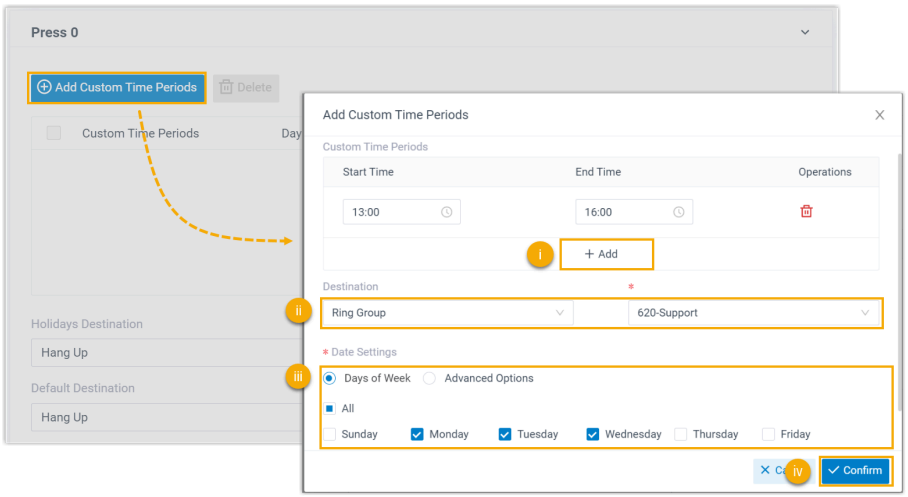
b. In the pop-up window, select the checkbox of **Time Condition**, and select **Same time condition for all key presses**.



c. In the **Time-based Mode** drop-down list, select **Based on Custom Time Periods**.



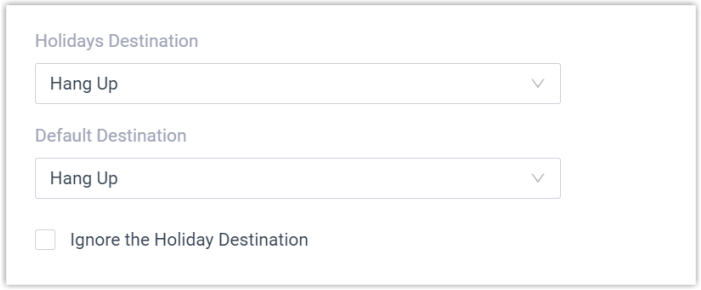
d. Under a key press event, click **Add Custom Time Periods** to create a custom time period for the key.



i. Click **Add** and specify the start time and end time.
ii. In the **Destination** drop-down list, specify the destination for calls within the custom time periods.
iii. In the **Date Settings** section, specify the workdays.
iv. Click **Confirm**.

> **Note:**
> The specified custom time periods ONLY take effect for this key press event.

e. Specify the following call routing destinations for the key press event.



- **Holidays Destination**: Specify the call routing destination for calls during holidays.

> **Note:**

> If you want to route incoming calls based on custom time periods during holidays, select the checkbox of **Ignore the Holiday Destination**.

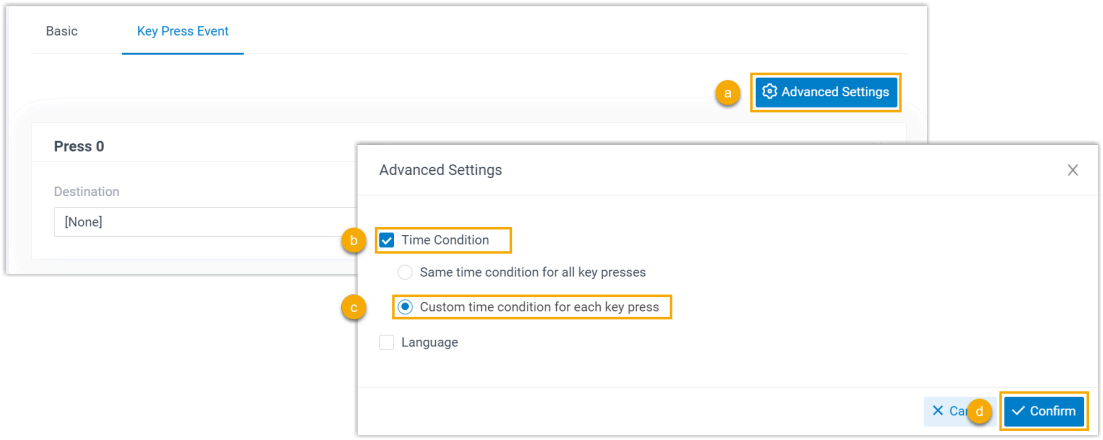
- **Default Destination**: Specify the call routing destination for calls outside the custom time periods.

f. Repeat step **d** - **e** for all the desired keys.

g. Click **Save** and **Apply**.

Apply different time conditions for each key press event

1. Log in to PBX web portal, go to **Call Features > IVR**.
2. Click beside the desired IVR.
3. In the **Key Press Event** tab, enable time-condition settings for this IVR.



a. On the top-right corner, click **Advanced Settings**

b. In the pop-up window, select **Time Condition**.

c. Select **Custom time condition for each key press**.

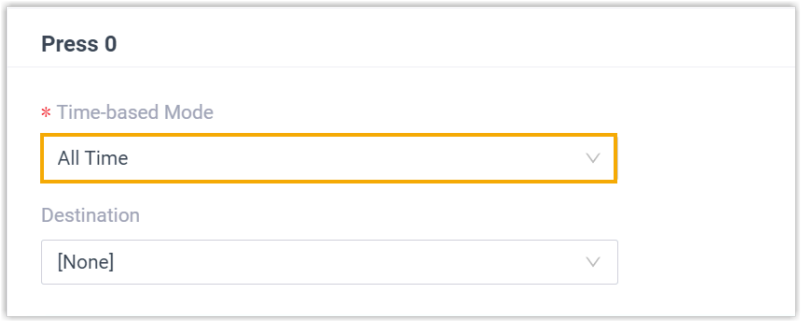
d. Click **Confirm**.

4. In the **Time-based Mode** drop-down list of a key, select the time condition and configure the call routing destinations.

Option	Description
All Time	Always route calls to the specified destination. For detailed instructions, see Route calls at all times.
Based on Global Business Hours	Route calls to specified destinations based on the global business hours, non-working hours, and holidays (Defined in **Call Control > Business Hours and Holidays**).

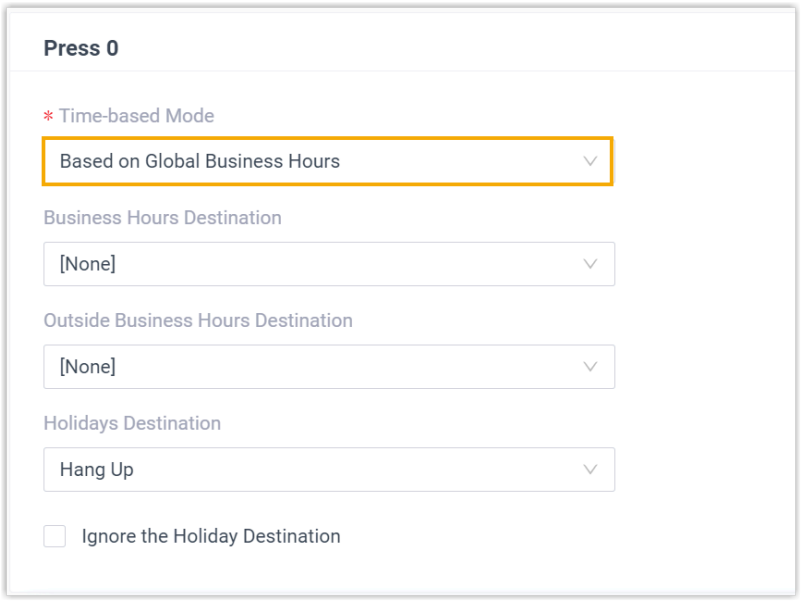
Option	Description
	For detailed instructions, see Route calls based on global business hours.
Based on Custom Business Hours	Create a custom business hours, and route calls to specified destinations based on the custom business hours, non-working hours, and holidays. For detailed instructions, see Route calls based on custom business hours.
Based on Custom Time Periods	Create a custom time period and route calls based on the custom time period. For detailed instructions, see Route calls based on custom time periods.

Route calls at all times



a. In the **Time-based Mode** drop-down list, select **All Time**.
b. In the **Destination** drop-down list, specify the destination.

Route calls based on global business hours

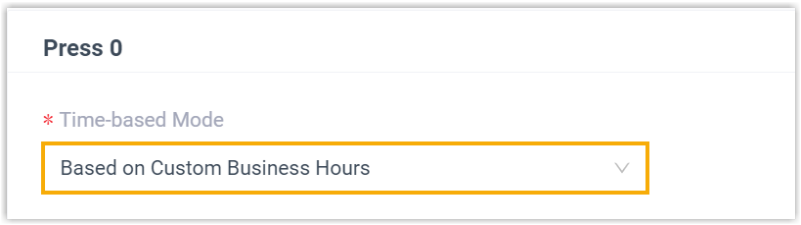


a. In the **Time-based Mode** drop-down list, select **Based on Global Business Hours**.
b. Specify call routing destinations for the key press event based on the global business hours.
    - **Business Hours Destination**
    - **Outside Business Hours Destination**
    - **Holidays Destination**

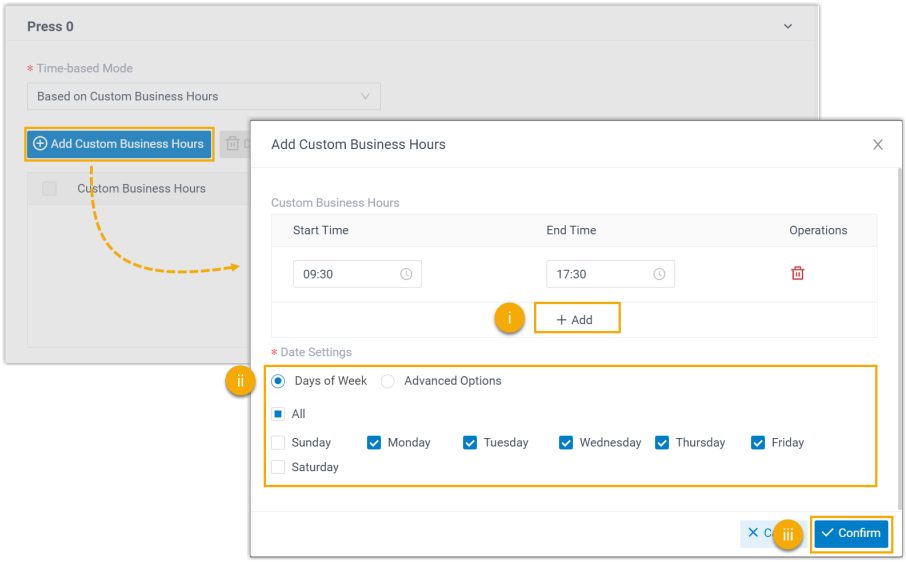
> **Note:**
> If you want to route incoming calls based on global business hours and non-working hours during holidays, select the checkbox of **Ignore the Holiday Destination**.

Route calls based on custom business hours

a. In the **Time-based Mode** drop-down list, select **Based on Custom Business Hours**.

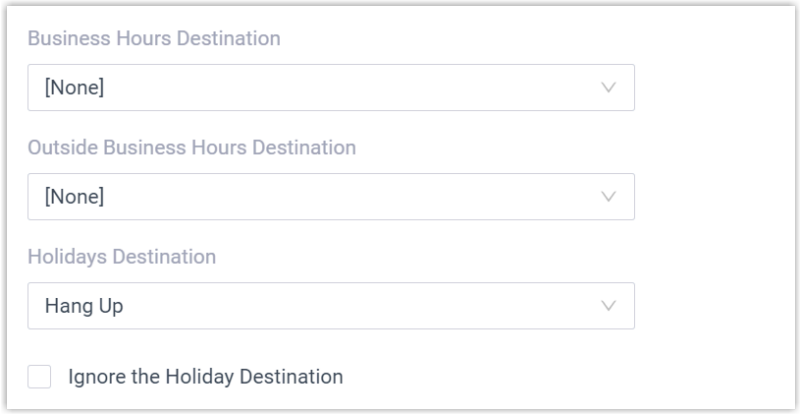


b. Click **Add Custom Business Hours** to create custom business hours.



i. Click **Add** and specify the start time and end time.
ii. In the **Date Settings** section, specify the workdays.
iii. Click **Confirm**.

c. Specify call routing destinations for a key press event based on the custom business hours.



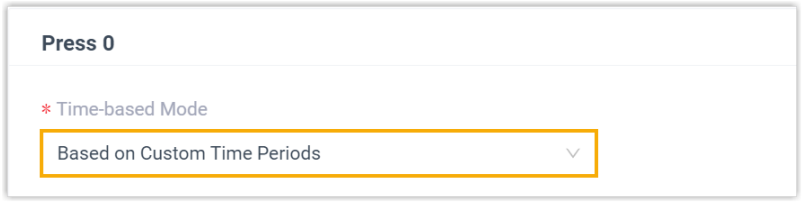
- **Business Hours Destination**

- **Outside Business Hours Destination**
- **Holidays Destination**

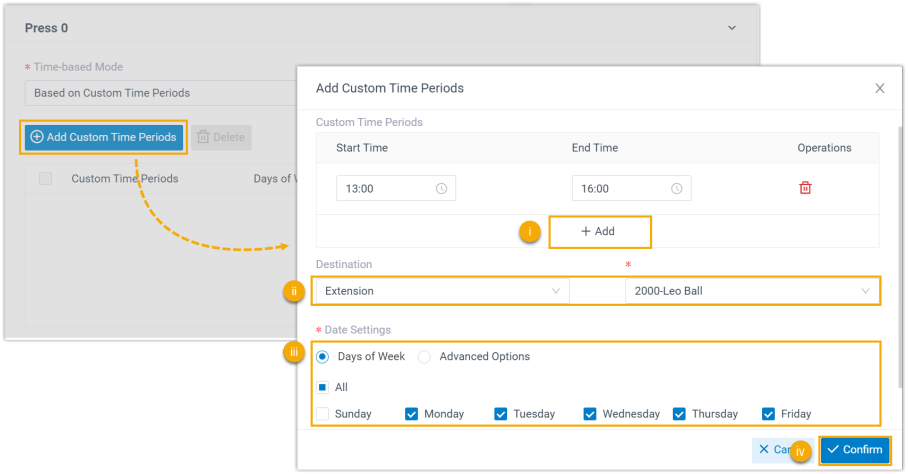
> **Note:**
> If you want to route incoming calls based on custom business hours and non-working hours during holidays, select the checkbox of **Ignore the Holiday Destination**.

Route calls based on custom time periods

a. In the **Time-based Mode** drop-down list, select **Based on Custom Time Periods**.

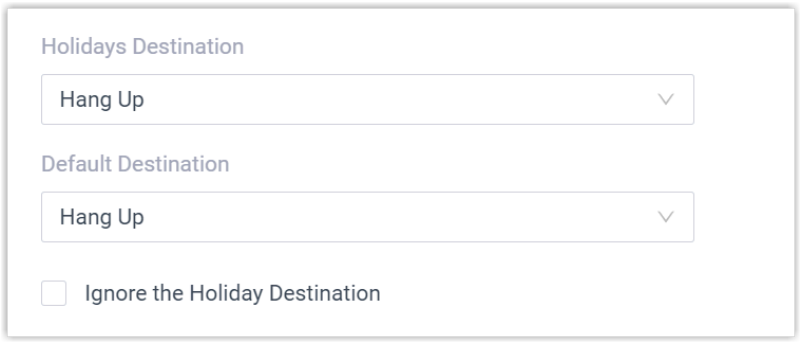


b. Click **Add Custom Time Periods** to create a custom time period.



i. Click **Add** and specify the start time and end time.
ii. In the **Destination** drop-down list, specify the destination for calls within the custom time periods.
iii. In the **Date Settings** section, specify the workdays.
iv. Click **Confirm**.

c. Specify the following call routing destinations for the key event.



- **Holidays Destination**: Specify the call routing destination for calls during holidays.

> **Note:**
> If you want to route incoming calls based on custom time periods during holidays, select the checkbox of **Ignore the Holiday Destination**.

- **Default Destination**: Specify the call routing destination for calls outside custom time periods.

5. Repeat step **4** for all the desired keys.
6. Click **Save** and **Apply**.

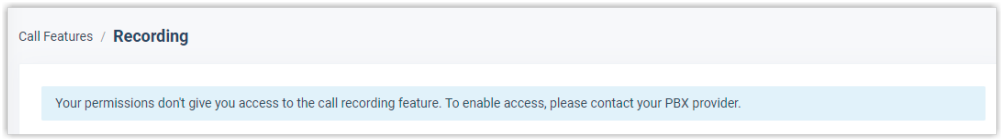
Call Recording

Call Recording Overview

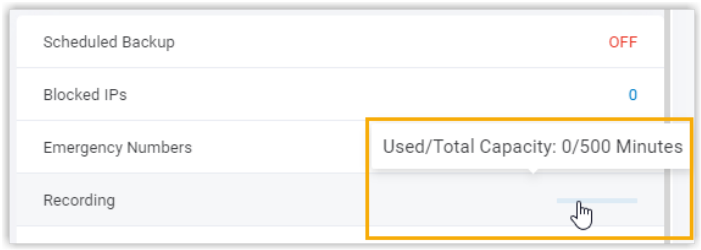
Call recording is valuable to keep important conversations, help train employees, evaluate their performance, and provide them with feedback. This topic describes how does call recording work, recording types, recording prompt, and recording management.

Background information

If there is a prompt on the top of the **Recording** configuration page, indicating that you have no permission to access this feature, you can contact your PBX provider to enable this feature if needed.



The recording file is stored in a default storage area. You can check the storage capacity at **Dashboard**.



The PBX will notify contacts concerned via the specified notification methods when the recording is about to reach the capacity limit. The oldest recording files will be automatically cleaned up when the storage is full.

How does call recording work

The system records the conversation automatically when a call is established. During call recording, the user can pause and resume recording to avoid the sensitive information being recorded. After the call ends, the system converts the conversation into audio files (.wav) with a digital signature.

> **Note:**
> The digital signature ensures a recording is not altered in any way.

Recording types

You can set up call recording for extensions, trunks, conferences, and queues respectively.

- **Extensions**: Record all the calls of the specified extensions, including the internal calls and external calls.

  > **Note:**
  > Paging/Intercom call and voicemail on the specified extension would not be recorded.

- **Trunks**: Record all the calls on the specified trunks, including inbound calls and outbound calls.

For example, for employees who use a dedicated trunk to deal with customer issues, the system only records all the calls on this trunk.

- **Conferences**: Record the conversation of all members who join the specified conference rooms.
- **Queues**: Record the calls based on the specified queues.

  For example, an agent logs in to two queues (Service and Support), and call recording is enabled for Service. The system can record all the calls from Service, but not record the calls from Support.

> **Note:**
> The system automatically records a queue call or a conference call only when you activate recording for a queue or conference. For example, extension 1000 is an agent of a queue, you activate recording for extension 1000, but not activate recording for the queue. When extension 1000 answers a queue call, the call is not recorded.

Recording prompts

By default, the system does not play any prompts when a call is being recorded.

To ensure that recordings are lawful and callers have given their consent, you can customize recording prompt for internal calls, inbound calls, and outbound calls respectively. The system plays the recording prompt before call recording begins.

Recording management

- **For users**: The users can monitor and switch call recording status on IP phones and Linkus Clients.
- **For administrator**: The administrator can manage the recording files, and grant permission to other users.

Set up Call Recording

This topic describes how to set up call recording for extensions, trunks, conferences, and queues.

Set up call recording for extensions

The system records the internal calls and external calls on the selected extensions.

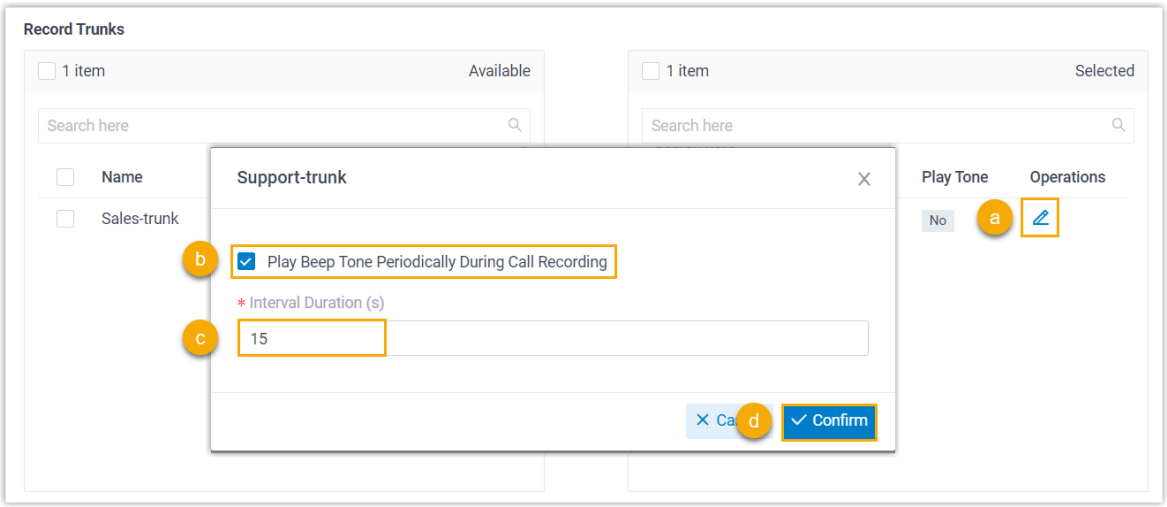
1. Log in to PBX web portal, go to **Call Features > Recording**.
2. **Optional:** Select the checkbox of **Enable Recording of Internal Calls** to automatically record the internal calls.
3. In the **Record Extensions** section, select the desired extensions from the **Available** box to the **Selected** box.
4. Click **Save** and **Apply**.

Set up call recording for trunks

The system automatically records the external calls on the selected trunks.

1. Log in to PBX web portal, go to **Call Features > Recording**.
2. In the **Record Trunks** section, select the desired trunks from the **Available** box to the **Selected** box.
3. **Optional:** To periodically play beep tones to inform both parties when the call on the selected trunk is being recorded, do as follows.

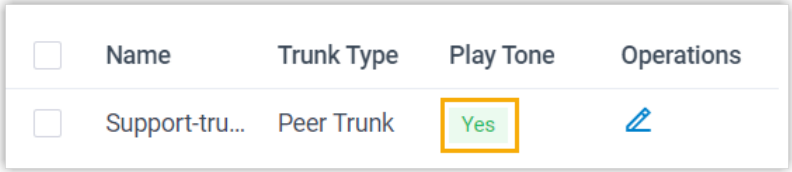
> **Note:**
> This setting will not take effect on the calls where multiple participants are involved, such as conference calls and multi-party calls.



a. In the **Selected** box, click beside a selected trunk.
b. In the pop-up window, select the checkbox of **Play Beep Tone Periodically During Call Recording**.
c. In the **Interval Duration (s)** field, specify the interval for playing the beep tone.

d. Click **Confirm**.

The **Play Tone** of the selected trunk displays **Yes**, indicating that periodic play-back of beep tone is enabled for the trunk.



4. Click **Save** and **Apply**.

Set up call recording for conferences

The system automatically records the calls on the selected conferences.

1. Log in to PBX web portal, go to **Call Features > Recording**.
2. In the **Record Conferences** section, select the desired conferences from the **Available** box to the **Selected** box.
3. Click **Save** and **Apply**.

Set up call recording for queues

The system automatically records the calls on the selected queues.

1. Log in to PBX web portal, go to **Call Features > Recording**.
2. In the **Record Queues** section, select the desired queues from the **Available** box to the **Selected** box.
3. Click **Save** and **Apply**.

Set up Recording Prompts

This topic describes how to set up recording prompts for internal calls, inbound calls, and outbound calls respectively.

Set up recording prompt for internal calls

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**, upload a custom prompt or record a custom prompt for internal calls.

Note:

> The uploaded file should meet the audio file requirements.

2. Go to **Call Features > Recording**.
3. In the **Internal Call Being Recorded Prompt** drop-down list, select a prompt for internal calls.
4. Click **Save** and **Apply**.

Set up recording prompt for inbound calls

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**, upload a custom prompt or record a custom prompt for inbound calls.

> **Note:**
> The uploaded file should meet the audio file requirements.

2. Go to **Call Features > Recording**.
3. In the **Inbound Call Being Recorded Prompt** drop-down list, select a prompt for inbound calls.
4. Click **Save** and **Apply**.

Set up recording prompt for outbound calls

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**, upload a custom prompt or record a custom prompt for outbound calls.

> **Note:**
> The uploaded file should meet the audio file requirements.

2. Go to **Call Features > Recording**.
3. In the **Outbound Call Being Recorded Prompt** drop-down list, select a prompt for outbound calls.
4. Click **Save** and **Apply**.

Allow Users to Switch Call Recording Status

By default, if you set up call recording for extensions, trunks, conferences, or queues, the specified calls would be recorded as soon as they are established, and all the users can NOT switch call recording status. To avoid sensitive information being recorded or to allow

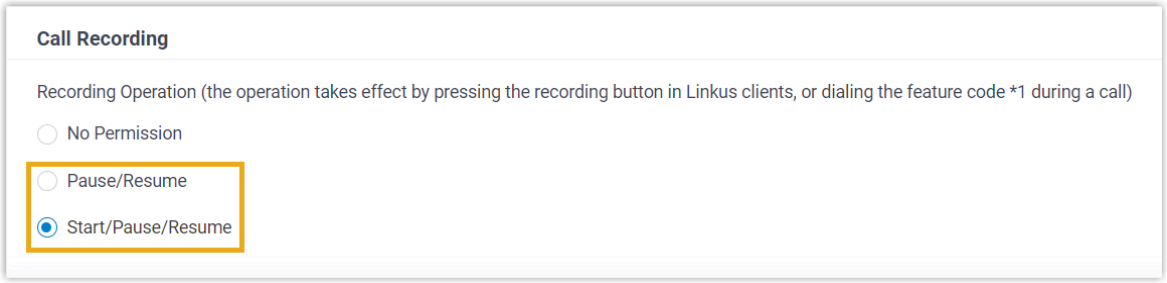
users to start recording their calls when necessary, you can grant permissions to specific users, so that they can start, pause, or resume recording during a call.

Restrictions

> **Note:**
> Extension users can NOT switch the recording status during a conference call, even if you have granted them permission.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit a desired extension.
2. Click **Features** tab.
3. In the **Call Recording** section, grant the operation permission of call recording to the extension user.



   - **Pause/Resume**: Allow the user to pause or resume recording during a call that is specified to be recorded.

     > **Tip:**
     > To specify the calls to be recorded, see Set up Call Recording.

   - **Start/Pause/Resume**: Allow the user to start, pause, or resume recording during any calls (except conference calls), be the calls specified to be recorded or not.
4. Click **Save** and **Apply**.

Result

- The user can switch call recording status in the following ways:
  - Press the recording button on Linkus Clients

◦ Dial a feature code

> **Note:**
> The default feature code for switching call recording status is `*1`.
> You can change, enable, or disable the code on PBX web portal (Path: **Call Features > Feature Code > Recording > Switch Extension's Recording Status**).

- By default, the user can view the call recording file that is generated when a call ends. To restrict the user from viewing recording files, see Restrict Users from Viewing Recording Files.

Monitor Call Recording Status on an IP phone

This topic describes how to set up a BLF key on an IP phone to monitor the call recording status.

Background information

For the users who want to know whether the call recording state is switched successfully or not, you can set a BLF key for each user by auto provisioning.

> **Note:**
> Users can also set function keys on their own IP phones. For more information, contact the phone manufacturer.

Procedure

1. Assign function keys for extension users to monitor agent status.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
      - If you want to assign function keys for a specific extension user, click beside the desired extension.
      - If you want to assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.
   b. Click the **Function Keys** tab.
   c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

- **Type**: Select **BLF**.
- **Value**: Enter the code (`*1`) followed by extension number (for example *11000).
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.
3. If the extension has been associated with a phone, reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.

Result

The BLF key shows the real-time status of call recording.

- **Red**: An active call of the monitored extension is being recorded.
- **Green**: The monitored extension is not in a call or the call recording is paused.
- **Off**: The BLF key does not subscribe the recording status of this extension. Check if your configurations are correct.

> **Note:**
> The key LED status may vary by phone models.

Manage Call Recording Files

This topic describes how to manage call recording files, including searching, playing, downloading, or deleting the recording files.

Search recording files

You can search the recording files by time, caller number, callee number, or call ID.

1. Log in to PBX web portal, go to **Reports and Recordings > Recording Files**.
2. Set the search criteria.
   - **Time**: Set the start date and the end date.

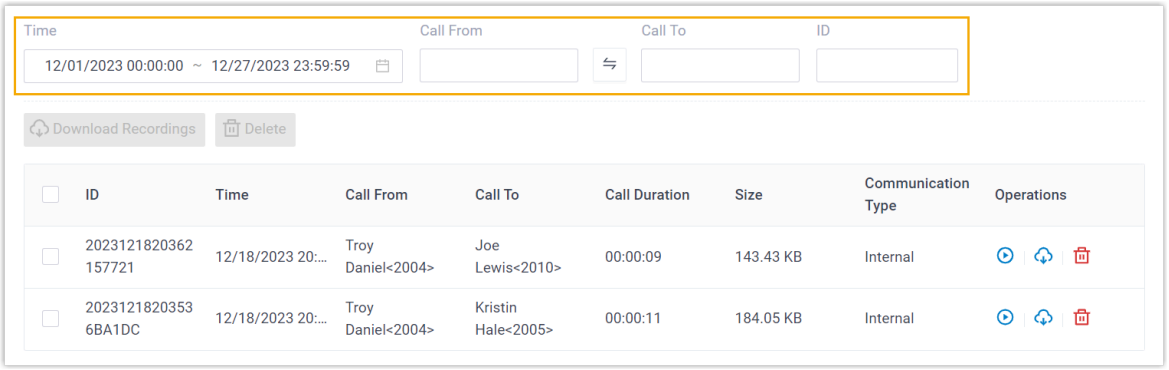
     To specify a time period, click **select time** to set the start time and the end time.
   - **Call From**: Set the caller's number or name.
   - **Call To**: Set the callee's number or name.

     > **Tip:**
     >
     > To swap the callee for the caller, click ⇆.
   - **ID**: Enter the unique identifier for the recording file.

   The search results are displayed in the list.

Play recording files

1. Log in to PBX web portal, go to **Reports and Recordings > Recording Files**.
2. **Optional:** Set the criteria to filter the desired recordings.



3. Click ▶ beside the recording to which you want to listen.
   - **Play on Web**: Click ▶ to play the call recording on the web directly.
   - **Play to Extension**: Play the call recording on the phone.
     a. Select an extension, and click **Play**.

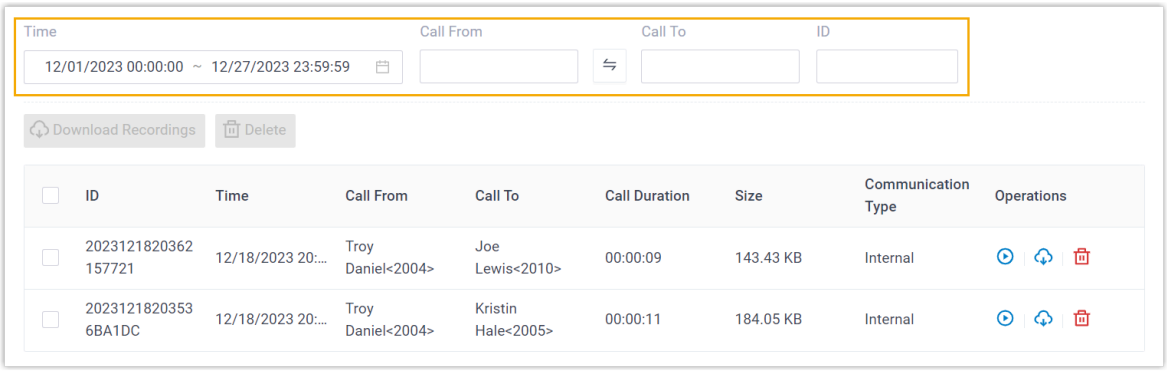
        The system places a call to the extension.
     b. Pick up the call to listen to the call recording on the phone.

Download recording files

You can download recording files to your local computer or to an external server (such as FTP server, SFTP server, Amazon S3, or Google Cloud Storage).

This section introduces how to download recording files to your local computer. For external server, see Yeastar P-Series Cloud Edition Remote Archiving Overview.

1. Log in to PBX web portal, go to **Reports and Recordings > Recording Files**.
2. **Optional:** Set the criteria to filter the desired recordings.

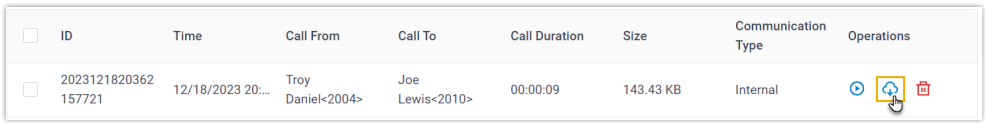


3. Download recording file(s) as needed.

> **Note:**
> You can download a maximum of 600 MB recording files or a maximum of 100 recording files at a time. The recordings that exceed the limit will not be downloaded.

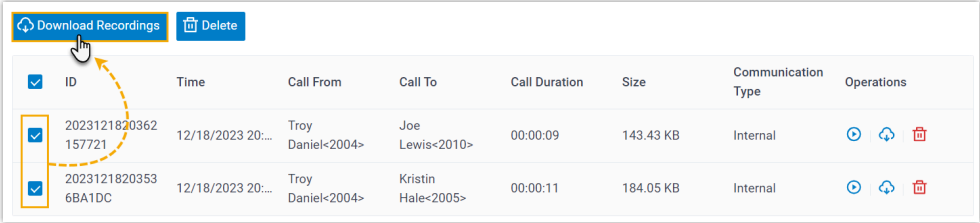
Download a recording file

Click beside the recording file.



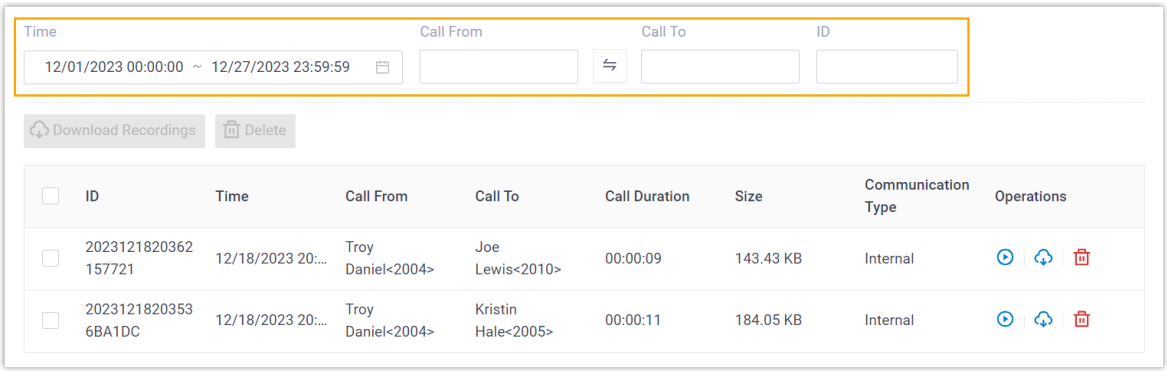
Download multiple recording files

Select the checkboxes of the recording files that you want to download, then click **Download Recordings**.



Delete recording files

1. Log in to PBX web portal, go to **Reports and Recordings > Recording Files**.
2. **Optional:** Set the criteria to filter the desired recordings.



3. Delete a recording file, or delete recording files in bulk.
    - **Delete a recording**: Select the recording file that you want to delete, click  and **OK**.
    - **Delete recordings in bulk**: Select the checkboxes of the recording files that you want to delete, click **Delete** and **OK**.

Grant Manage Permission of Recording Files

By default, only the super administrator has permission to manage the call recording files. This topic describes how to grant manage permission to extension users.

Background information

As a super administrator, you can grant manage permission to a role, and assign the role to extension users. When the user logs in to the web client, he/she can manage recording files.

Procedure

1. Set up a user role.
    a. Log in to PBX web portal, go to **Extension and Trunk > Role**, edit a role.
    b. In the **Reports and Recordings** section, specify the manageable extensions and accessible permissions of recordings files for the role.
        - **Manage Extensions**: Specify the manageable extension range.
        - **Recording Files Operation Permission**: Specify the accessible permission, including **Play**, **Download**, and **Delete**.
    c. click **Save**.
2. Assign a role to a user.
    a. Go to **Extension**, edit the extension to which you want to grant recording permission.
    b. In the **User Information** section, select the role from the **User Role** drop-down list.
    c. Click **Save** and **Apply**.

Restrict Users from Viewing Recording Files

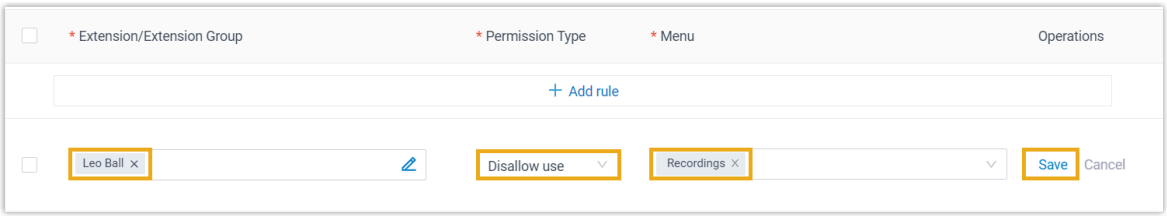
By default, all the users have access to viewing their own recording files. For security reasons, you can restrict specific users from view recording files.

Requirements

Server / Client	Version Requirement
PBX Server	Version 84.12.0.32 or later
Linkus Mobile Client	• **Linkus iOS Client**: Version 5.2.9 or later • **Linkus Android Client**: Version 4.13.16 or later
Linkus Desktop Client	• **Windows Desktop**: Version 1.2.14 or later • **Mac Desktop**: Version 1.2.10 or later

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Client Permission > Menu Visibility**.
2. Click **Add rule** to create a menu visibility rule.
3. Set up the rule, then click **Save**.



- **Extension/Extension Group/Organization**: Click to select desired extensions, extension groups, or departments, for which you want to restrict the viewing permission of recordings.
- **Permission Type**: Select **Disallow use**.
- **Menu**: Select **Recordings**.

Result

The extension user can not view recording files on his or her Linkus UC Clients.

Ring Group

Ring Group Overview

Ring group is a feature to share the distribution of incoming calls among employees. This topic describes what is ring group, ring strategy, failover destination, and missed call alerts.

What is ring group

Ring group allows you to merge multiple extension numbers into a virtual number. The customers can dial the virtual number, and the calls ring through all the members to make sure that no call goes unanswered. It is often used to efficiently distribute calls to specific departments such as Sales, Support, and Accounting.

Ring strategy

Ring group can ring members in three ways:

- **Ring all simultaneously**: When receiving an incoming call, the system rings all the available members at the same time and stops ringing when any member in the group picks up the call. If no one answers the call within the ring time, the system routes the call to the failover destination.

- **Ring sequentially**: When receiving an incoming call, the system rings the first available member in the list. If no answer within the ring time, the system rings the next available member until the last one. If no one answers the call, the system routes the call to the failover destination.
- **Memory hunt**: When receiving an incoming call, the system rings the first available member in the list. If no answer within the ring time, the system rings the first and second available member. If still no answer within the ring time, the system rings the first, second, and third available member, and the like, until all available members in the list rang. If no one answers the call, the system routes the call to the failover destination.

Failover destination

When a call comes in to the ring group, and no one is available to answer the call, you can end the call or route the call to the following destinations:

- **Hang Up**
- **Extension**
- **Extension Voicemail**
- **Group Voicemail**
- **IVR**
- **Ring Group**
- **Queue**
- **External Number**
- **Play Prompt and Exit**

Missed call alerts

When there are missed calls from ring group, the system can record the missed calls and notify members via email.

To record missed calls from ring group, see Record Missed Calls.

To set up email alerts for missed calls from ring group, see Set up Email Notifications for Missed Calls.

Create a Ring Group

This topic describes how to create a ring group.

Procedure

1. Log in to PBX web portal, go to **Call Features > Ring Group**, click **Add**.
2. Configure the ring group.
   - **Number**: Enter a virtual number for callers to access the group.

     The default ring group number range is from 6300 to 6399.
   - **Name**: Enter a group name to help you identify it.
   - **Ring Strategy**: Select a ring method to distribute calls to members.
     - **Ring All**: Ring all available extensions simultaneously.
     - **Ring Sequentially**: Ring all available extensions sequentially.
     - **Memory Hunt**: Ring the first available extension in the list. If no answer within the ring time, progressively add the next available extension to ring, until all the available extensions in the list rang.
   - **Ring Timeout (s)**: Set a number of seconds that the system waits before ringing next member or routing the call to **Failover Destination**.
   - **Welcome Prompt**: Optional. Specify a welcome prompt that will be played to callers before members answer the call.

     You can select an existing prompt from the drop-down list, or click **Record New** / **Upload** to record or upload a new prompt.

     > **Note:**
     > The existing prompts are synchronized from **PBX Settings > Voice Prompt > Custom Prompt**. If you record or upload a new prompt here, it will also be synchronized to the location.
   - **Music on Hold**: Optional. Specify the music that will be played to callers when the call is put on hold.

     You can select an existing MoH playlist from the drop-down list, or click **Create New** to add a new one.

     > **Note:**
     > The existing playlists are synchronized from **PBX Settings > Voice Prompt > Music on Hold**. If you create a new playlist here, it will also be synchronized to the location.
   - **Ring Group Alert Info**: Optional. Set an "alert info text" to add to Alert-info header in INVITE request for ring group calls.

When receiving a ring group call, the phone will inspect "Alert-Info" header to determine which ring tone it should use for ringing.

- **Members**: Select the desired extensions from the **Available** box to the **Selected** box.
- **Failover Destination**: Select a destination to route the call when no member answers the call within ring time.
  - **Hang up**: End the current call.
  - **Extension**: Route the call to the specified extension.
  - **Extension Voicemail**: Route the call to voicemail box of the specified extension.
  - **Group Voicemail**: Route the call to voicemail box of a queue, a ring group, or a custom group.
  - **IVR**: Route the call to the specified IVR.
  - **Ring Group**: Route the call to another ring group.
  - **Queue**: Route the call to the specified queue.
  - **External Number**: Route the call to an external number.
  - **Play Prompt and Exit**: Play a custom prompt, and then hang up the call.
- **Record Missed Calls**: Decide whether to record missed calls from ring group.

> **Note:**
> - This option is available only when both **Ring Strategy** and **Failover Destination** are set to the followings:
>   - **Ring Strategy** is set to **Ring All** or **Memory Hunt**.
>   - **Failover Destination** is set to **Extension**, **Queue**, or **Ring Group**.
> - You can also set up email alert on missed calls from ring group for members. To achieve this, enable this option, then turn on email notifications on missed calls for members. For more information, see Set up Email Notifications for Missed Calls.

3. Click **Save** and **Apply**.

What to do next

Set up an inbound route, and specify the destination to the queue.

Manage Ring Groups

This topic describes how to edit a ring group, and delete ring groups.

Edit a ring group

You can edit the group settings, including adding or removing a member, or change the ring strategy.

1. Log in to PBX web portal, go to **Call Features > Ring Group**.
2. Click  beside the ring group that you want to edit.
3. Change the ring group settings according to your needs.
4. Click **Save** and **Apply**.

Delete ring groups

1. Log in to PBX web portal, go to **Call Features > Ring Group**.
2. To delete a ring group, click  beside the ring group that you want to delete.
3. To delete ring groups in bulk, select the checkboxes of the ring groups that you want to delete, click **Delete**.
4. Click **OK** and **Apply**.

Call Queue

Call Queue Overview

Call queue is a method of handling large calls and provides callers with engaging holding experiences. This topic describes what is call queue, queue compositions, queue preference, and call center service.

What is call queue

A queue is like a virtual waiting room, in which callers wait in line to talk with the available agent. When the customer calls in PBX and reaches the queue, he/she can hear the hold music and announcement while the queue distributing the call to the available agents.

What is Call Center service

Call Center service is an additional service that drives faster call resolution and real-time call center monitoring, reporting, and management. It provides a powerful call center console, including a customizable Wallboard for proactive tracking of 16 key performance metrics, and a switchboard-type Queue Panel for real-time monitoring & control of queue activities, insightful call center reports, SLA and more.

Configuration guide

For more detailed information and configurations of call queue and Yeastar Call Center, see Call Center Administrator Guide.

Feature Code

Configure Feature Codes

Feature codes are a set of digits that the extension user can dial to activate a specific feature. This topic describes how to configure feature codes.

Background information

Yeastar P-Series Cloud Edition provides various feature codes for users to activate or deactivate a specific feature. You can change, enable, or disable the code, and change the digit timeout for entering the feature code.

For more information about feature code, see Feature Code Reference.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. In the **Feature Code Digit Timeout (ms)** field, enter a number of seconds for inputting next digit.

   The digit timeout is the time between consecutive key presses on the phone's keypad.
3. Decide whether to enable or disable a feature code.
   - **Enable a feature code**: Select the checkbox of the specific feature code.
   - **Disable a feature code**: Unselect the checkbox of the specific feature code.
4. **Optional:** Change the code according to your needs.
5. Click **Save** and **Apply**.

Feature Code Reference

This topic describes the list of default feature codes.

Recording

Name	Default Code	Usage
Switch Extension's Recording Status	*1	• An extension user with **Pause/Resume** recording operation permission dials *1 to pause or resume recording during a call that is specified to be recorded. • An extension user with **Start/Pause/Resume** recording operation permission dials *1 to start, pause, or resume recording during any calls (except conference calls), be the calls specified to be recorded or not.

Call Flip

Name	Default Code	Usage
Call Flip	*01	Dial *01 to flip an active call from current device to another device.

Voicemail

Name	Default code	Usage
Check Voicemail/Subscribe Voicemail Status	*2	• To check the voicemail of extension 1000, dial *21000. • To check the group voicemail of queue 6400, dial *26400.
Leave a Voicemail for Extension/Group Voicemail	*12	• To leave a voicemail message for extension 1000, dial *121000. • To leave a voicemail message for queue 6400, dial *126400.

Call Transfer

Name	Default code	Usage
Attended Transfer	*3	Press *31000 to attended transfer a call to extension 1000.

Name	Default code	Usage
Blind Transfer	*03	Press *031000 to blind transfer a call to extension 1000.
Rebound Missed Transferred Calls (Applicable to Both blind/Semi-attended Transfer)	/	If enabled, when an extension user performs a blind transfer/semi-attended transfer and the transfer recipient doesn't answer, the call will be transferred back to the extension user.

Call Forwarding

Name	Default code	Usage
Enable "Forward All Calls"	*31	• Dial *31 to forward all calls to one's own voicemail. • Dial *311000 to forward all calls to extension 1000.
Disable "Forward All Calls"	*031	• Dial *031 to disable the automatic call forwarding of all calls.
Enable "Forward When Busy"	*32	• Dial *32 to forward calls to one's own voicemail when busy. • Dial *321000 to forward calls to extension 1000 when busy.
Disable "Forward When Busy"	*032	• Dial *032 to disable the automatic call forwarding when busy.
Enable "Forward No Answer"	*33	• Dial *33 to forward the no-answered calls to one's own voicemail. • Dial *331000 to forward the no-answered calls to extension 1000.
Disable "Forward No Answer"	*033	• Dial *033 to disable the automatic call forwarding of no-answered calls.

Call Pickup

Name	Default code	Usage
Group Call Pickup	*4	Dial *4 to pick up the ringing call for a group member.
Extension Pickup	*04	Dial *041000 to pick up the ringing call for extension 1000.

Call Parking

Name	Default code	Usage
Call Parking	*5	Dial *5 during a call to park a call.
Directed Call Parking	*05	Dial *056000 during a call to park a call to parking number 6000.

Call Monitoring

Name	Default code	Usage
Listen	*51	An authorized user dials *511001 to listen to the call of extension 1001 in real time. The authorized user can NOT talk with both parties.
Whisper	*52	An authorized user dials *521001 to listen to the call of extension 1001 in real time. The authorized user can talk with extension 1001 without being heard by the other party.
Barge-in	*53	An authorized user dials *531001 to listen to the call of extension 1001 in real time. The authorized user can talk with both parties.

Force Drop

Name	Default code	Usage
Call Force Drop	*54	• An authorized user dials *541001 to force drop the current call of extension 1001. • An authorized user dials *54 while monitoring the call of extension 1001 to force drop the call.

Intercom

Name	Default code	Usage
Intercom	*6	Dial *61001 to place an intercom call to extension 1001.

Queue

Name	Default code	Usage
Log in/Log out	*7	A dynamic agent dials *76400 to log in to or log out of queue 6400.
Pause/Unpause	*07	Dial *076400 to pause or unpause calls from queue 6400.

Hot Desking

Name	Default code	Usage
Guest In	*84	A user can dial *84 on a hot desking phone to register his or her extension to the phone.
Guest Out	*084	A user can dial *084 on a hot desking phone to de-register his or her extension from the phone.

Busy Camp-on

Name	Default code	Usage
Enable Busy Camp-on	*79	A user dials *791000 to camp on extension 1000 in case he or she gets a busy signal when dialing extension 1000. The PBX will automatically call the user back when extension 1000 becomes available.
Disable Busy Camp-on	*079	A user dials *079 to cancel all Busy Camp-on requests that he or she has initiated.

Speed Dial

Name	Default code	Usage
Speed Dial Prefix	*89	Specify a number to speed dial code 1, dial *891 to dial the specified number.

Presence Status

Name	Default code	Usage
Available	*91	Dial *91 to switch one's own presence status to **Available**.
Away	*92	Dial *92 to switch one's own presence status to **Away**.

Name	Default code	Usage
Do Not Disturb	*93	Dial *93 to switch one's own presence status to **Do Not Disturb**.
Lunch Break	*94	Dial *94 to switch one's own presence status to **Lunch Break**.
Business Trip	*95	Dial *95 to switch one's own presence status to **Business Trip**.
Off Work	*96	Dial *96 to switch one's own presence status to **Off Work**.

Switch Business Hours and Holidays Status

Name	Default code	Usage
Keep the Business Hours Status or the Time Condition after Switching	/	If enabled, keep the status after users switched the PBX's Business Hours status or dialed a feature code to switch the time condition of an inbound route.
Switch Global Business Hours and Holidays Status	*99	Dial *99 to override time condition for Global Business Hours and Holidays.
Time Condition Switching Prefix	*8	Dial a feature code starting with *8 to switch the time condition of inbound routes that are based on Custom Business Hours or Custom Time Periods.

Boss-Secretary Feature

Name	Default code	Usage
Enable Boss-Secretary Feature	*59	**Note:** This feature code is only available for extensions set as the boss extension. Dial *59 to enable the Boss-Secretary feature.
Disable Boss-Secretary Feature	*059	**Note:** This feature code is only available for extensions set as the boss extension. Dial *059 to disable the Boss-Secretary feature.

Pause Reason

Yeastar P-Series Cloud Edition provides queue agents with the following default pause reasons and the corresponding feature codes. You can modify the default settings, or add new ones.

> **Note:**
>
> - Yeastar P-Series Cloud Edition supports up to 20 feature codes for pause reasons.
> - The feature codes are synchronized with the settings on **Call Features > Queue > Pause Reason**.

After you set the feature codes, agents can dial feature codes with a format of `Pause feature code` + `queue number` + `pause reason feature code` to pause with a specific reason.

Pause reason	Default code	Usage
Lunch	*01	An agent can dial *076400*01 to pause receiving calls from queue 6400 for lunch.
Break	*02	An agent can dial *076400*02 to pause receiving calls from queue 6400 for a break.
Wrap up	*03	An agent can dial *076400*03 to pause receiving calls from queue 6400 for after-call processing.

Conference

Conference Overview

Conference calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings. This topic describes what is conference call, and conference member.

What is conference call

Yeastar P-Series Cloud Edition supports dial-in conference that allows multiple participants, including internal users and external users, to start a conference call, and talk to each other anywhere and anytime.

Conference member

- **Moderator**: The conference moderator is a participant who can lock the conference call and manage the participants in a conference call.
- **Participant**: The conference member who can talk with each other and adjust the volume.

Create a Conference Room

To make a conference call, you should create a conference room on Yeastar P-Series Cloud Edition first. This topic describes how to create a conference room.

Procedure

1. Log in to PBX web portal, go to **Call Features > Conference**, click **Add**.
2. Set up the conference room.
    - **Number**: Enter a room number for callers to dial into the conference call.
    - **Name**: Enter a room name to help you identify it.
    - **Participant Password**: Optional. The participants need to enter the password to join conference call.
    - **Moderator Password**: Optional. The participants can enter the password to join conference call as moderators.
    - **Voice Prompt**: Select a prompt to announce to the participants when someone joins or exits from the conference call.
        - **Default**: Prompt a tone when participant joins or exits from conference call.
        - **Extension**: Prompt the extension number of the participant when the participant joins or exits from conference call.
    - **Wait for Moderator**: Whether to forbid the participants to talk with each other till the moderator joins the conference call.
    - **Allow Extension Participants to Invite**: Whether to allow the extension participants to invite users to join the conference.
    - **Moderator(s)**: Select the moderators.

      The moderators can join the conference calls without any password.
3. Click **Save** and **Apply**.

What to do next

If the external participants want to join conference, you need to set an inbound route and specify the **Destination** to **Conference**.

Join a Conference Call

Both the PBX extension users and the external users can join the conference. This topic describes how to join a conference call.

Join as a conference participant

1. Dial the conference room number.
2. If participant password is required, enter the participant password.

   If you are the first participant in the conference call, the system plays a hold music to you.

Join as a conference moderator

For moderators

If you are a moderator specified by administrator, you can dial the conference room number to join the conference call.

If you are the first participant in the conference call, the system plays a hold music to you.

For participants who want to join conference as moderators

If you are not a moderator, and the moderator password is set for the conference room, you can also join conference call as a moderator

1. Dial the conference room number.
2. Enter the moderator password.

   If you are the first participant in the conference call, the system plays a hold music to you.

Invite Users to a Conference Call

By default, only the conference moderators can invite users to the conference. This topic describes how to allow participants to invite users and how to invite users to a conference call.

Allow participants to invite users

1. Log in to PBX web portal, go to **Call Features > Conference**, edit the desired conference.
2. Select the checkbox of **Allow Extension Participants to Invite**.
3. Click **Save** and **Apply**.

Invite users to a conference call

1. During a conference call, press the # key.

   You are forced out of the conference call temporarily.
2. Dial the number that you want to invite.

   After the invited user joins or rejects the conference call, you will return to the conference call.

Manage Conference Rooms

This topic describes how to edit conference room settings and delete conference rooms.

Edit a conference room

> **Note:**
> You can not change the conference room number after setting up a conference room.

1. Log in to PBX web portal, go to **Call Features > Conference**.
2. Click beside the conference room that you want to edit.
3. Change the conference room settings according to your needs.
4. Click **Save** and **Apply**.

Delete conference rooms

You can delete a conference room, or delete conference rooms in bulk.

1. Log in to PBX web portal, go to **Call Features > Conference**.
2. To delete a conference room, do the following:

   a. Click beside the conference room that you want to delete.

b. Click **OK** and **Apply**.

3. Delete conference rooms in bulk, do the following:
   a. Select the checkboxes of the conference rooms that you want to delete, click **Delete**.
   b. Click **OK** and **Apply**.

Conference Voice Menu

This topic describes the conference voice menu.

During the conference call, the participants can manage the conference by pressing * key on their phones to access voice menu for conference room.

The following table shows the conference voice menu.

Key	Description	Moderator	Participant
1	Mute or unmute yourself.	√	√
2	Lock or unlock the conference.	√	×
3	Eject the last user.	√	×
4	Decrease the conference volume.	√	√
6	Increase the conference volume.	√	√
7	Decrease your volume.	√	√
8	Exit the voice menu.	√	√
9	Increase your volume.	√	√

Speed Dial

Speed Dial Overview

Speed dial is often the easiest way to quickly connect with people and extensions that you dial frequently. This topic describes what is speed dial, and how to use speed dial.

What is speed dial

Speed dial is a feature that allows you to assign a speed dial code to a number that the users frequently dial. When dialing long strings of overseas numbers, the users do not have to remember or enter long telephone numbers on their phones.

How to use speed dial

You can create speed dial with a **Prefix** in front of the **Speed Dial Number** to avoid interference with your extensions.

- **Speed Dial Number**: The shorter number you assign to the phone number.
- **Prefix**: The code to access the speed dial feature. The default prefix is *89.

The users can dial {*prefix*}+{*speed_dial_number*} to call an assigned phone number. For example, assign `1` to phone number `5503302`, dial `*891` to place a call to `5503302`.

Set up Speed Dial Prefix

You need to dial the speed dial code with a prefix. The prefix is used to access the speed dial feature, and avoid interference with the extensions. This topic describes how to set up speed dial prefix.

Procedure

1. Log in to PBX web portal, go to **Call Features > Speed Dial**.
2. Click **Prefix**.
3. In the **Speed Dial Prefix** field, enter a prefix according to your needs.

   > **Note:**
   > - Only numbers and special characters `*` `#` are allowed.
   > - The maximum character length is 7.

   The default speed dial prefix is `*89`.
4. Click **Save** and **Apply**.

> **Tip:**
> To disable the speed dial prefix, go to **Call Features > Feature code > Speed Dial > Speed Dial Prefix**.

Add a Speed Dial Number

This topic describes how to add a speed dial number.

Background information

- Assume that you have an outbound route that allows you to dial an external number 15990234988, and you want to dial speed number 111 to reach an external number 15990234988 through the route.
- The speed dial prefix is enabled and set to `*89`.

Procedure

1. Log in to PBX web portal, go to **Call Features > Speed Dial**, click **Add**.
2. In the **Speed Dial Number** field, enter `111`.
3. In the **Phone Number** field, enter `15990234988`.
4. Click **Save** and **Apply**.

Result

Dial *89111 on your phone to call the external number 15990234988.

Manage Speed Dial Numbers

This topic describes how to edit a speed dial number, or delete speed dial numbers.

Edit a speed dial number

1. Log in to PBX web portal, go to **Call Features > Speed Dial**.
2. Click  beside the speed dial entry that you want to edit.
3. Change the **Speed Dial Number** or **Phone Number** according to your needs.
4. Click **Save** and **Apply**.

Delete speed dial numbers

You can delete a speed dial entry, or delete speed dial entries in bulk.

1. Log in to PBX web portal, go to **Call Features > Speed Dial**.
2. To delete a speed dial number, do as follows:
    a. Click  beside the speed dial entry that you want to delete.
    b. Click **OK** and **Apply**.
3. To delete speed dial numbers in bulk, do as follows:

a. Select the checkboxes of the speed dial entries that you want to delete, click **Delete**.
b. Click **OK** and **Apply**.

Export and Import Speed Dial Numbers

The speed dial numbers configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired speed dial numbers in the exported file, and import the file to PBX again. This topic describes how to export and import speed dial numbers.

Export speed dial numbers

You can export all speed dial numbers to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Features > Speed Dial**.
2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Speed Dial Number Parameters.

Import speed dial numbers

We recommend that you export speed dial numbers to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8 .CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Speed Dial Number Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Features > Speed Dial**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.

4. Click **Import**.

   The speed dial numbers in the CSV file will be displayed in the **Speed Dial** list.

Related information

Import and Export -FAQ

Call Transfer

Call Transfer Overview

Call transfer is an in-call feature that allows the users to transfer current calls from their phones to another phone number or extension. This topic describes the call transfer types, and call transfer options.

Call transfer types

There are two scenarios to transfer a call:

- **Attended Transfer**: An attended transfer, also called consult transfer or warm transfer, allows the transferor to consult with the transfer recipient before transferring a call, such as the assistant can confirm with the executive whether he is free to answer the call before transferring the call.
- **Blind Transfer**: A blind transfer, also called cold transfer, allows the transferor to transfer a call to transfer recipient immediately without consultative communication, such as transfer a call to ring group.

Call transfer options

The following options are available for you to set up call transfer:

- **Feature code**: Extension users can use the call transfer code to transfer a call.

  The default call transfer code:

  - **Attended Transfer**: *3
  - **Blind Transfer**: *03
- **Digit Timeout(s)**: The timeout for transferor to enter the transfer recipient's number after dialing the feature code. The time interval between each digit should be within the digit timeout.

- **Attended Transfer Timeout(s)**: The ring timeout for transfer recipient to take the transferring call.

  If the transfer recipient does not answer the transferring call within the timeout, the system sends the call back to transferor.
- **Rebound Missed Transferred Calls (Applicable to Both blind/Semi-attended Transfer)**: If enabled, when an extension user performs a blind transfer/semi-attended transfer and the transfer recipient doesn't answer, the system sends the call back to the extension user.

Set up Call Transfer

This topic describes how to set up call transfer.

Set up attended transfer

1. Log in to PBX web portal, go to **Call Features > Feature code > Call Transfer**.
2. Select the checkbox of **Attended Transfer** to enable the attended transfer feature.

   If unselected, the extension users can not perform attended transfer by dialing the feature code.
3. Enter a code number according to your needs.
4. In the **Digit Timeout(s)** drop-down list, select a timeout for entering transfer recipient's number after you hear a dial tone.
5. In the **Attended Transfer Timeout(s)** field, enter a number of seconds for transfer recipient to take the transferring call.
6. Click **Save** and **Apply**.

Set up blind transfer

1. Log in to PBX web portal, go to **Call Features > Feature code > Call Transfer**.
2. Select the checkbox of **Blind Transfer** to enable the blind transfer feature.

   If unselected, the extension users can not perform blind transfer by dialing the feature code.
3. Enter a code number according to your needs.
4. In the **Digit Timeout(s)** drop-down list, select a timeout for entering transfer recipient's number after you hear a dial tone.

5. In the **Rebound Missed Transferred Calls (Applicable to Both blind/Semi-attended Transfer)** option, set whether to transfer the call back to the transferor when the transfer recipient doesn't answer the call.
6. Click **Save** and **Apply**.

Perform an Attended Transfer

If you want to make sure someone is ready to take a transferred call or you need to explain something to the transfer recipient, you can perform an attended transfer. This topic describes how to perform an attended transfer.

Procedure

1. During a call, press the feature code of attended transfer (default *3).

   The original call is placed on hold, and the system prompts "transfer" and the dial tone.
2. Dial the phone number of the contact where you want the call to be transferred.
3. Wait for the call to be answered.

   When the call is answered, talk to the transfer recipient.
4. Hang up the call directly to complete the call transfer.

   The original caller and the transfer recipient are connected.

Perform a Blind Transfer

If you do not need any interaction with the user who receives the call, you can perform a blind transfer. This topic describes how to perform a blind transfer.

Procedure

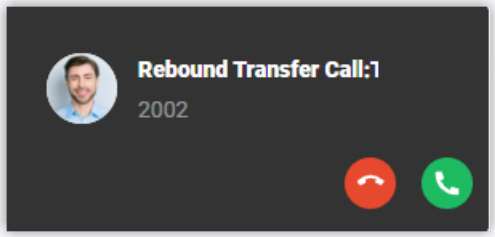
1. During a call, press the feature code of blind transfer (default *03).

   The original call is placed on hold, and the system prompts "transfer" and the dial tone.
2. Dial the phone number of the contact where you want the call to be transferred.

   The call ends automatically, and the transfer recipient's phone rings.

   A new call between original caller and transfer recipient is established after transfer recipient answers.

Note:
By default, if the transfer recipient doesn't answer the call, the call will be routed to the call forwarding destination of the transfer recipient. If you have enabled **Rebound Missed Transferred Calls (Applicable to Both blind/Semi-attended Transfer)**, the call will be transferred back to you with Caller ID displayed as **Rebound Transfer Call:** *{caller_id_name}*.



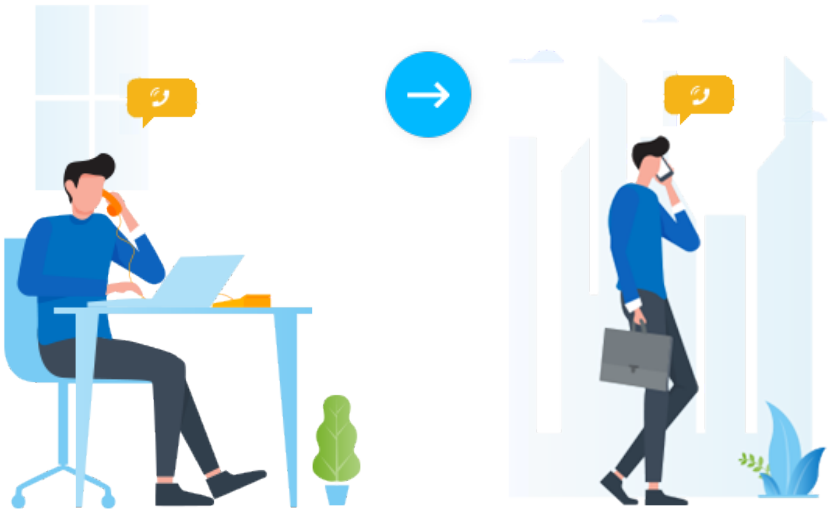
Call Flip

Call Flip Overview

Call Flip feature allows users to flip their ongoing calls from current device to another (with their extensions registered), without any interruption to the conversation.

Scenario

Assume that a sales representative is in a call with a customer on the desk phone, but has to get out of the office. In this case, the sales representative can flip the call to his mobile phone, keeping talking without customer knowing the switchover.



Methods of Call Flip

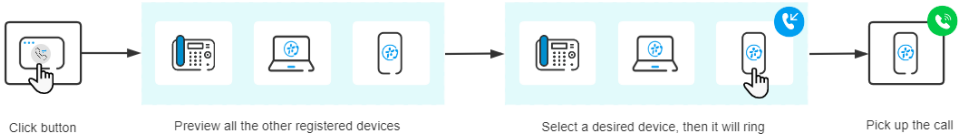
Extension users can flip an active call in the following ways:

- Flip an active call by clicking 'Call Flip' button
- Flip an active call by dialing 'Call Flip' feature code

Flip an active call by clicking 'Call Flip' button

With a simple click of the **Call Flip** button, users can preview all the other devices where their extensions are registered, and select a device to flip the call.

We provide a flowchart to help you understand the workflow:



This method is supported on the following endpoints:

- **Linkus Mobile Client**

  For more information, see Flip an Active Call between Devices.
- **Linkus Desktop Client**

  For more information, see Flip an Active Call between Devices.
- **Linkus Web Client**

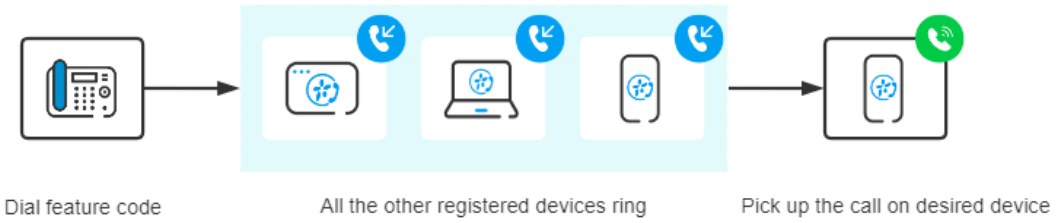
  For more information, see Flip an Active Call between Devices.

> **Note:**
> If Linkus Web Client is associated with 'Yeastar Linkus for Google', see Flip an Active Call between Devices.

Flip an active call by dialing 'Call Flip' feature code

By dialing the **Call Flip** feature code, all the other devices where users' extensions are registered will simultaneously ring. Users pick up the call on a desired device, then the call would be flipped.

We provide a flowchart to help you understand the workflow:



With the **Call Flip** feature code enabled on Yeastar P-Series Cloud Edition, this method is supported on all endpoints.

For more information about how to enable the **Call Flip** feature code, see Enable 'Call Flip' feature code.

For more information about how to flip an active call by dialing the feature code, see Flip an Active Call by Dialing a Feature Code.

Enable or Disable 'Call Flip' Feature Code

This topic describes how to enable or disable 'Call Flip' feature code.

Enable 'Call Flip' feature code

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. In the **Call Flip** section, select the checkbox, then configure the feature code.



3. Click **Save** and **Apply**.

   During a call, extension users can dial the feature code to flip the active call to another device where their extensions are registered. For more information, see Flip an Active Call by Dialing a Feature Code.

Disable 'Call Flip' feature code

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. In the **Call Flip** section, unselect the checkbox.



3. Click **Save** and **Apply**.

   Extension users can NOT flip an active call by dialing the feature code.

Flip an Active Call by Dialing a Feature Code

This topic describes how to flip an active call from current device to another (with the same extension registered) by dialing a feature code.

Requirements

- **PBX Server**: 84.8.0.25 or later
- **Extension**: Extension has been registered on more than one device.

Procedure

1. During an active call, dial the **Call Flip** feature code (default: `*01`).

   All the other devices where the extension is registered simultaneously ring.
2. Answer the call on a desired device.

Result

The call is flipped to the device, and the rest of the devices stop ringing.

Call Pickup

Call Pickup Overview

Call Pickup is a feature that allows employees to pick up colleagues' calls remotely, without having to walk to the his/her telephone. This topic describes the two pickup types including extension call pickup, group call pickup, and pickup code.

Extension call pickup

Extension call pickup, also known as directed call pickup, allows employees to pick up a call for a specific extension.

For example, the executive's phone is ringing, and the assistant knows the executive is in a meeting and is unavailable to answer the call, the assistant can pick up the executive’s call from his/her phone.

Group call pickup

Group call pickup allows extension group members to share their incoming calls. For a group of employees working on the same subject, when a member receives an incoming call and is unavailable to take the call, other members can pick up the call from their phones.

Note:

- If the extension group has multiple ringing calls at the same time, the first ringing call will be picked up.

- The extension group does NOT include the group that contains all the PBX extensions.

Pickup feature code

Extension users can use the pickup code to pick up a call.

The default pickup code:

- **Group Call Pickup**: *4
- **Extension Pickup**: *04

Tip:
You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Pickup**.

Pick up a Call for a Group Member

This topic describes how to set up a Feature key on an IP phone to pick up a call for an extension group member.

Background information

For the users who want to pick up a call for an extension group member, you can set a Feature key for each user by auto provisioning. Users can also set function keys on their own IP phones. For more information, contact the phone manufacturer.

Note:
The default feature code for picking up a group member's call is `*4`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Pickup > Group Call Pickup**.

Set up a Feature key

The following takes Yealink phone as an example to set a Speed Dial key for group pickup.

1. Assign function keys for extension users to monitor extension status.

a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.

- If you want to assign function keys for a specific extension user, click beside the desired extension.
- If you want o assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.

b. Click the **Function Keys** tab.

c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

- **Type**: Select **Speed Dial**.
- **Value**: Enter the code (*4).
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.
3. If the extension has been associated with a phone, reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.

Result

- When extension group members receive a call, the user can press the Feature key directly to answer the call.
- When the call is picked up, the IP phones of other extension group members display a missed call.

Pick up a Call for a Specific Extension

This topic describes how to set up a BLF key on an IP phone to pick up a call for a specific extension.

Background information

For the users who want to monitor call status changes of a specific extension, and pick up the call on their phones, you can set a BLF key for each user by auto provisioning. Users can also set function keys on their own IP phones. For more information, contact the phone manufacturer.

> **Note:**
> The default feature code for picking up an extension call is `*04`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Pickup > Extension Pickup**.

Set up a BLF key

The following takes Yealink phone as an example to set a BLF key for call pickup.

1. Assign function keys for extension users to monitor agent status.
   a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
      - If you want to assign function keys for a specific extension user, click beside the desired extension.
      - If you want to assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.
   b. Click the **Function Keys** tab.
   c. Configure function keys.

      > **Note:**
      > The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

      - **Type**: Select **BLF**.
      - **Value**: Enter the code (`*04`) followed by extension number (for example *041001).
      - **Label**: Optional. Enter a value, which will be displayed on the phone screen.
   d. Click **Save**.

2. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.
3. If the extension has been associated with a phone, reprovision the phone to take effect.
    a. Go to **Auto Provisioning > Phones**.
    b. Click beside the phone assigned to this extension.

Result

- When the monitored extension receives an incoming all, the BLF key fast flashes red. The user can press the BLF key directly to answer the call.
- When the call is picked up, the IP phone where the monitored extension is registered displays a missed call.

Call Parking

Call Parking Overview

Call parking is a method of holding a call on a phone, anyone can retrieve the call on another phone. This topic describes what is call parking, parking number, parking types, parking recall, and parking code.

What is call parking

Traditionally, you can only retrieve the call on the same phone when you hold a call. Call parking allows you to hold a call on a parking number, and allows you to dial the parking number on any phone to retrieve the call.

Parking number

Parking number, also known as slot or orbit, is a virtual extension number that the system assigns to the parked call. One parked call occupies one parking number.

The maximum number of simultaneous parking number is 100.

Parking types

Yeastar P-Series Cloud Edition supports two parking types.

- **Call parking**: Park a call randomly on the first available parking number.

- **Directed call parking**: Park a call on the specified parking number.

Parking timeout destination

The parked call remains on the parking number for a specified period of time (default 60 seconds). If no one retrieves the parked call within the timeout period, the system routes the call to a designated destination (default initiator).

Parking feature code

Extension users can use the parking code to park a call.

The default parking code:

- **Call Parking**: *5
- **Directed Call Parking**: *05

> **Tip:**
> You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Parking**.

Directed Call Parking

This topic describes how to park a call on a specific parking number, and retrieve the parked call.

Background information

For sales or support, it probably doesn't matter exactly who picks up the call. You can allocate different parking numbers to different departments. For example, 6099 for sales, 6098 for support, and so on. The receptionist can park the call directly on the parking number based on business. Anyone in the department can retrieve the call by the parking number.

> **Note:**
> The default range of parking number is from 6000 to 6099. The randomly call parking occupies parking number from 6000. To avoid that the allocated parking number is occupied by randomly call parking, we recommend that you allocate the parking number backwards from 6099.

Prerequisites

Make sure that the parking number is vacant. If the specified parking number is occupied, the system parks the call to the next available parking number.

Tip:
Set up a function key for users to monitor the status of parking number.

- For receptionist, he/she can press the function key to park the call to the parking number.
- For users in different departments, a parked call is visible on the function key, so that they can press the function key to retrieve the parked call easily.

Procedure

Parking number 6099 is assigned to salesmen. The receptionist receives a call, and the customer wants to consult business information.

1. The receptionist dials *056099 to park the call to parking number 6099.
2. The receptionist tells the sales there is a parked call for business.

   If function keys are configured on the sales' IP phones, they will be notified.
3. The sales who is available can dial 6099 or press the function key to retrieve the call.

Tip:
The default feature code for directed call park is `*05`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Parking > Directed Call Parking**.

Call Parking

This topic describes how to park a call randomly on the available parking number, and retrieve the parked call.

Background information

During a conversation, the employee may need to go to another office for retrieving an important file or for security, he/she can park the call, and to continue the conversation after arriving at the other office.

Procedure

1. Dial the feature code (*5) to park a call.

   The system prompts you the parking number (6000) where the call is parked.
2. Go to another office, dial the parking number (6000) to retrieve the parked call.

> **Tip:**
> The default feature code for call park is `*5`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Call Parking > Call Parking**.

Set up Parking Number

The default range of parking number is from 6000 to 6099. This topic describes how to define the range of parking numbers for parked call.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature code > Call Parking**.
2. In the **Parking Number Range** field, enter a number range for parked call.
3. Click **Save** and **Apply**.

> **Tip:**
> You can also change the parking number range at **PBX Settings > Preferences > Extension Preference > Parking Extension**.

Set up Parking Timeout Destination

By default, if a parked call is not retrieved after 60 seconds, the call will be transferred back to the originator. You can set up the parking timeout and timeout destination. This topic describes how to set up parking timeout and timeout destination for an unretrieved call.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code > Call Parking**.
2. In the **Parking Timeout (s)** field, enter the number of seconds for the parked call.

3. In the **Timeout Destination** drop-down list, select a destination to receive the unretrieved call.

   A parked call will be routed to the designated destination when the call parking times out.

   - **Call Parking Initiator**: Route the call to the user who parks this call.
   - **Extension**: Route the call to the designated extension number.
   - **Extension Voicemail**: Route the call to the designated extension's voicemail.
   - **Group Voicemail**: Route the call to the voicemail box of a queue, a ring group, or a custom group.
   - **External Number**: Route the call to the designated external number.

   > **Note:**
   > Set the **Prefix** according to your outbound route so that PBX can successfully route incoming calls to external number.
   > - If the **Strip** of outbound route is not set, you don't have to set the **Prefix**.
   > - If the **Strip** of outbound route is set, you need to set the **Prefix** according to the **Patterns** of outbound route.

4. Click **Save** and **Apply**.

Set up Music on Hold for Call Parking

This topic describes how to customize music on hold for call parking on Yeastar P-Series Cloud Edition.

By default, the system plays a default music on hold (set on **PBX Settings > Voice Prompt > Prompt Preferences > Music on Hold**) to the parked party when a call is parked. You can refer to this topic to customize the music for call parking.

Requirements

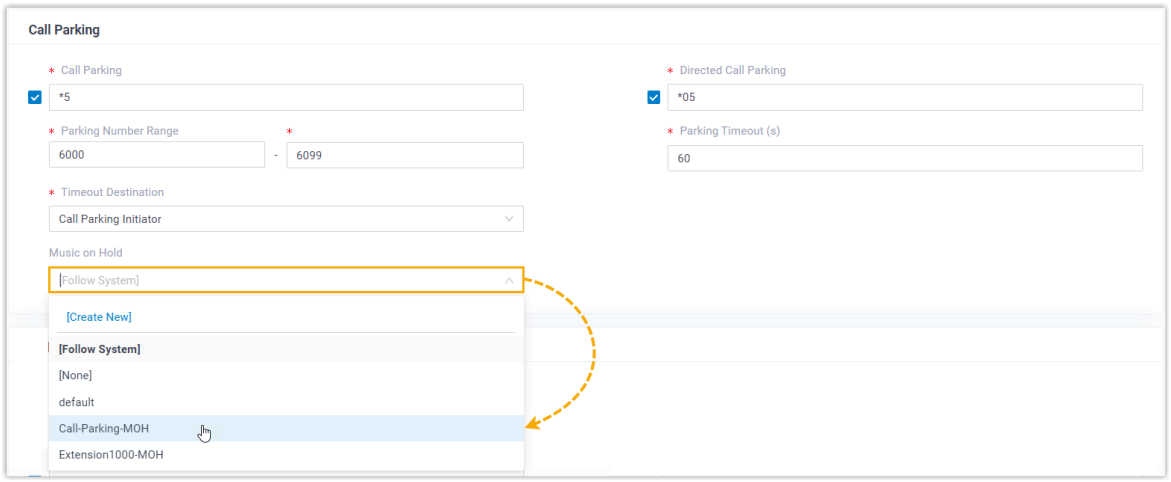
The firmware version of PBX server is 84.15.0.22 or later.

Prerequisites

You have prepared and uploaded a custom Music on Hold (MoH) playlist.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code > Call Parking**.
2. In the **Music on Hold** drop-down list, select the MoH playlist that you have uploaded.



3. Click **Save** and **Apply**.

Monitor a Parking Number on an IP Phone

This topic describes how to set up a function key on a user's phone to monitor a parking number.

Background information

For users who use directed call parking and want to monitor a specific parking number, you can set a function key for each user by auto provisioning.

> **Note:**
> Users can also set function keys on their own IP phones. For more information, contact the phone manufacturer.

> **Tip:**
> Agents can also press the function keys to park or retrieve a call.

Procedure

1. Assign function keys for extension users to monitor parking number.

a. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.

- If you want to assign function keys for a specific extension user, click beside the desired extension.
- If you want o assign function keys for multiple extensions, select the checkboxes of the desired extensions, and click **Edit**.

b. Click the **Function Keys** tab.

c. Configure function keys.

> **Note:**
> The number of programmable keys varies by phone models. If the number of function keys you assign to an extension exceeds the number of programmable keys, the redundant function keys cannot take effect.

- **Type**: Select **Park & Retrieve**.
- **Value**: Select a parking number.
- **Label**: Optional. Enter a value, which will be displayed on the phone screen.

d. Click **Save**.

2. If the extension hasn't been associated with a phone, see Auto Provision IP Phones to bind a phone with the extension.
3. If the extension has been associated with a phone, reprovision the phone to take effect.
   a. Go to **Auto Provisioning > Phones**.
   b. Click beside the phone assigned to this extension.

Result

The function key shows the real-time status of the parking number.

- **Green**: The parking number is idle.

  The user can press the function key to park an active call to the idle parking number.
- **Red**: The parking number is occupied.

  The user can press the function key to retrieve a parked call from the monitored parking number.

> **Note:**

The key LED status may vary by phone models.

Call Monitoring

Call Monitoring Overview

Call monitoring feature allows you to listen in on employee calls without interference or joining in the conversation as a third party. It helps you check on the quality of teams' sales calls, learn more about customer reactions and insights, and gain a better view for coaching and training the team.

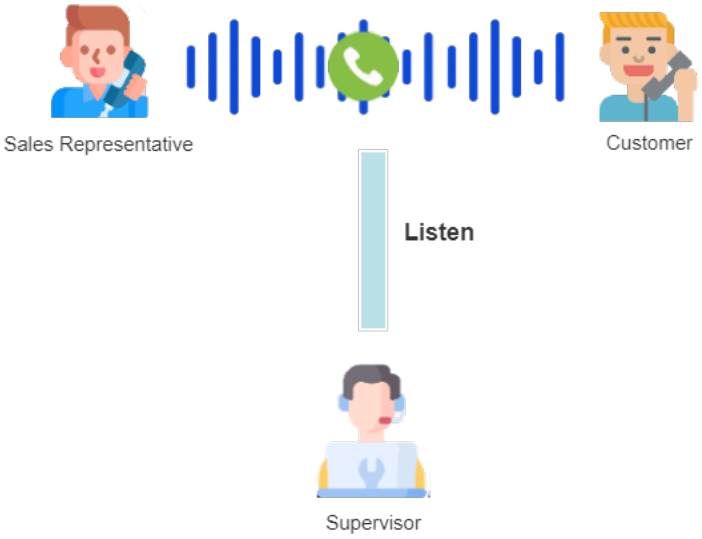
Modes of call monitoring

Yeastar P-Series Cloud Edition supports the following monitoring modes:

Listen mode

Listen mode allows the authorized user to listen in on a call in real time, but can NOT talk with either party.

This mode is often used for supervisor to track the daily actions of sales representatives and evaluate their performance on the sales process.



Whisper mode

Whisper mode allows the authorized user to listen in on a call in real time, and directly talk with the monitored extension without being heard by the other party.

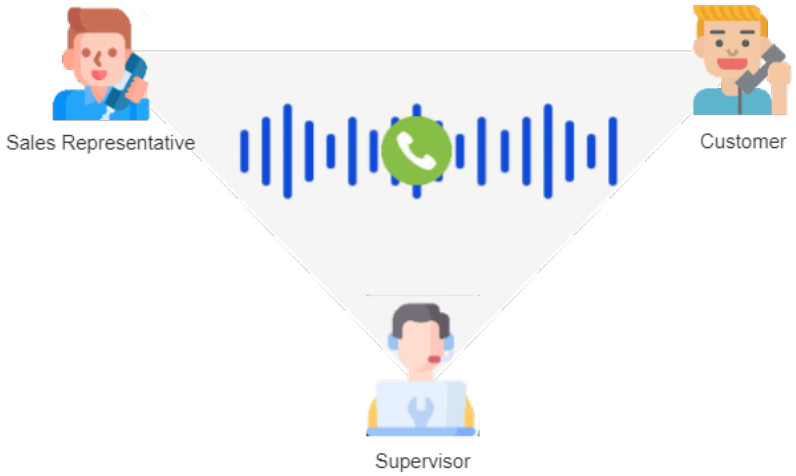
This mode is often used for supervisor to coach remote sales representatives and train them as they work on a sales call.



Barge-in mode

Barge-in mode allows the authorized user to listen in on a call in real time, and talk with both parties.

This mode is often used for supervisor to help sales representatives handle a difficult objection and move deals forward.



Methods of call monitoring

An authorized user can listen in on a call in the following ways:

- Operator Panel
- Queue Panel
- Dial **Feature Code + Extension Number**

Operator Panel

For users who have access to Operator Panel, you can assign the permission of call monitoring operations to them. In this way, the authorized users can listen in on an active call using any one of the three monitoring modes when they are working on Operator Panel.

To assign the permission of call monitoring operations on Operator Panel, see View or Change Permissions for Group Members.

To listen in on a call on Operator Panel, see Monitor a Call.

Queue Panel

For queue managers who have access to Queue Panel, you can assign the permission of call monitoring operations to them. In this way, the queue managers can listen in on an active call using any one of the three monitoring modes when they are working on Queue Panel.

To assign the permission of call monitoring operations on Queue Panel, see Grant Queue Panel Permissions.

To listen in on a call on Queue Panel, see Monitor a Call.

Dial 'Feature Code + Extension Number'

For users who only have phones on hand, you can configure feature codes for each call monitoring mode, then assign permission to specific users. In this way, the authorized users can listen in on an active call using the specified monitoring mode by dialing **Feature Code + Extension Number** on their phones.

To configure the feature code and assign the permission of call monitoring operations, see Allow Users to Monitor a Call by Dialing a Feature Code.

Allow Users to Monitor a Call by Dialing a Feature Code

If users only have phones on hand, you can configure feature codes for each monitoring mode, then assign permission to users. In this way, the authorized users can listen in on a call by dialing a feature code on their phones.

Background information

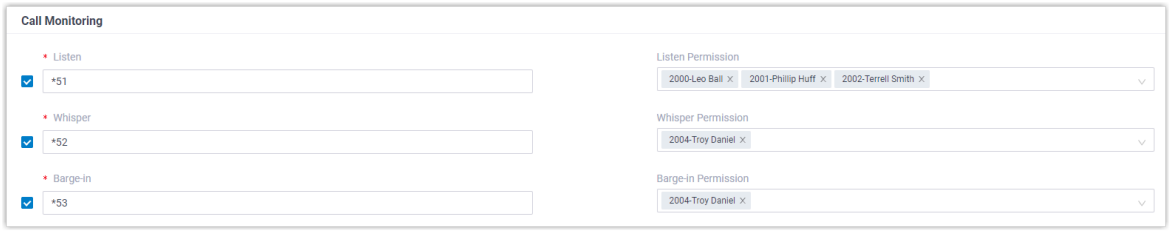
By default, all the extension users can NOT monitor others' calls by dialing a feature code, but their calls can be monitored instead.

To allow specific extension users to monitor others' calls by dialing a feature code, follow the procedure shown below.

To prevent specific extension users from being monitored, see Disallow Users to be Monitored by Others.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. In the **Call Monitoring** section, configure the feature code and assign permission to users.



   a. Select the checkbox of a desired call monitoring mode, then configure the feature code.
      - **Listen**: Listen in on a call in real time, but can NOT talk with either party.

        The default feature code is *51.
      - **Whisper**: Listen in on a call in real time, and directly talk with the monitored extension without being heard by the other party.

        The default feature code is *52.
      - **Barge-in**: Listen in on a call in real time, and talk with both parties.

        The default feature code is *53.
   b. In the **Listen/Whisper/Barge-in Permission** drop-down list, select the allowed extensions for each call monitoring mode respectively.

3. Click **Save** and **Apply**.

Result

The authorized user can dial **Feature Code + Extension Number** to monitor the calls of the extensions that are allowed to be monitored.

Important:

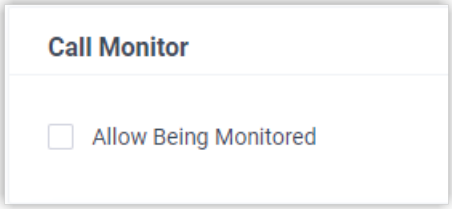
- Monitoring **Conference Calls** via feature code is NOT supported.
- Monitoring calls of extensions that are invisible on Linkus clients is NOT supported.
- During an internal call where one party allows being monitored while the other party disallows, the call can NOT be monitored even by the authorized user.

Disallow Users to be Monitored by Others

By default, all the extension users are allowed to be monitored. To keep specific users' calls private, you can disable the monitoring feature for users.

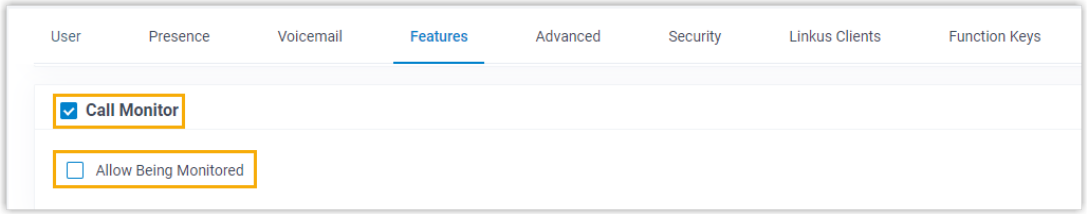
Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. To prevent an extension from being monitored, do as follows:
   a. Click beside the desired extension.
   b. Click **Features** tab.
   c. In the **Call Monitor** section, unselect the checkbox of **Allow Being Monitored**.



   d. Click **Save** and **Apply**.
3. To prevent multiple extensions from being monitored, do as follows:
   a. Select the checkboxes of the desired extensions, then click **Edit**.
   b. Click **Features** tab.

c. Select the checkbox of **Call Monitor**, then unselect the checkbox of **Allow Being Monitored**.



d. Click **Save** and **Apply**.

Result

The users' calls can NOT be monitored by anyone.

Force Drop

Allow Users to Force Drop Extensions' Calls

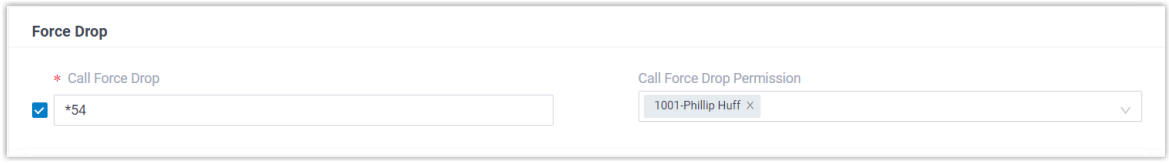
Call Force Drop feature allows the authorized users to force disconnect an extension's current call by dialing a feature code.

Requirements

The version of PBX is 84.16.0.25 or later.

Procedure

1. Log in to PBX web portal, go to **Call Features > Feature Code**.
2. Scroll down to the **Force Drop** section, then complete the following settings.



   - **Call Force Drop**: Retain the default feature code `*54` or change it as needed.
   - **Call Force Drop Permission**: Select the extensions that are allowed to force drop extensions' calls by dialing the feature code.
3. Click **Save** and **Apply**.

Result

The authorized users can dial *{feature_code}* + *{extension_number}* to force drop an extension's current call, or dial *{feature_code}* to force drop the monitored extension's current call.

For more information, see Force Drop an Extension's Call.

Force Drop an Extension's Call

This topic describes how an authorized user can force drop an extension's current call by dialing a feature code.

Prerequisites

The user to force drop calls has been assigned the **Call Force Drop** permission.

Procedure

- The authorized user dials *{feature_code}* + *{extension_number}* to force drop an extension's current call.

  For example, dial `*541000` to force drop the current call of extension 1000.
- The authorized user dials *{feature_code}* to force drop the monitored extension's current call.

  For example, while monitoring the current call of extension 1000, dial `*54` to force drop the call.

Result

The extension's current call is disconnected, and each user is prompted as follows:

- The authorized user who forces drop the call hears a prompt "Call force drop succeeded".
- The extension user whose call is forcibly disconnected hears a prompt "This call was forcibly dropped".
- The other party on the call hears a busy tone.

Boss-Secretary

Set up Boss-Secretary Feature for Extensions

Yeastar Boss-Secretary feature allows the secretary to filter incoming calls for the boss. You can enable this feature on an extension to designate it a boss extension, then assign secretary extension(s) to it and set forwarding rule. After the setup, any incoming calls to the boss extension that meet the forwarding rule will be automatically forwarded to the secretary extension. The secretary can answer the call and decide whether to transfer it to the boss.

Video Tutorial

Prerequisites

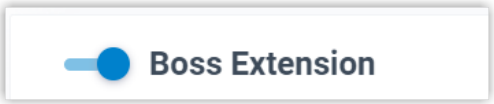
- If you want to assign multiple secretary extensions to one boss extension, you need to group all the secretary extensions into one ring group.

  For more information, see Create a Ring Group.

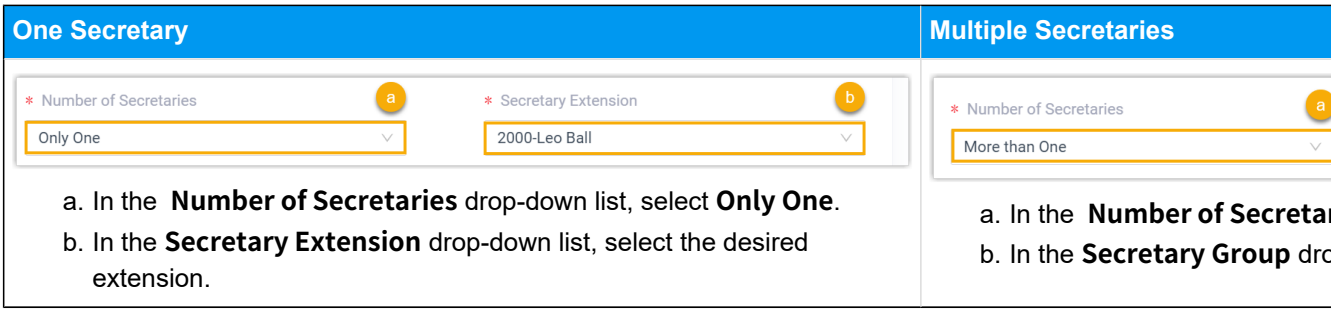
- If you want the boss extension to ring simultaneously when calls are forwarded to the secretary, you need to place both the boss extension and the secretary extension(s) into one ring group.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click beside the extension that you want to set as the 'Boss Extension'.
3. Click the **Features** tab.
4. In the **Boss Extension** section, turn on the switch to set the extension as the boss extension.

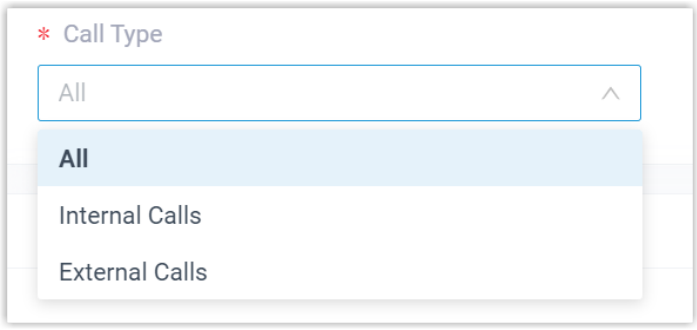
   The preselected **Enable Boss-Secretary Feature** option indicates the feature is enabled for this extension.
5. Specify secretary extension(s) for this boss extension.



a. In the **Number of Secretaries** drop-down list, select **Only One**.
b. In the **Secretary Extension** drop-down list, select the desired extension.

a. In the **Number of Secretar**
b. In the **Secretary Group** dro

6. In the **Call Type** drop-down list, decide which incoming calls to the boss extension need forwarding to the secretary extension(s).



- **All**: Forward all the incoming calls firstly to the secretary extension(s).
- **Internal Calls**: Forward only the internal incoming calls firstly to the secretary extension(s).
- **External Calls**: Forward only the external incoming calls firstly to the secretary extension(s).

> **Note:**
> If you set up Call Handling Based on Caller ID for the boss extension, the incoming calls that meet the call handling rule will be handled accordingly, instead of following the forwarding rule you set here.

7. Click **Save** and **Apply**.

Result

The extension is set up with the Boss-Secretary feature and designated as the boss extension.

The following features are achieved:

- Incoming calls of the selected call type to the boss extension will be automatically forwarded to the secretary extension(s). The secretary can answer the call and determine if it needs to be transferred to the boss.
- A call can be transferred between the boss and secretaries by putting it on hold and resuming it.
  When one party holds a call, the other party can resume and answer the call on the following endpoints:

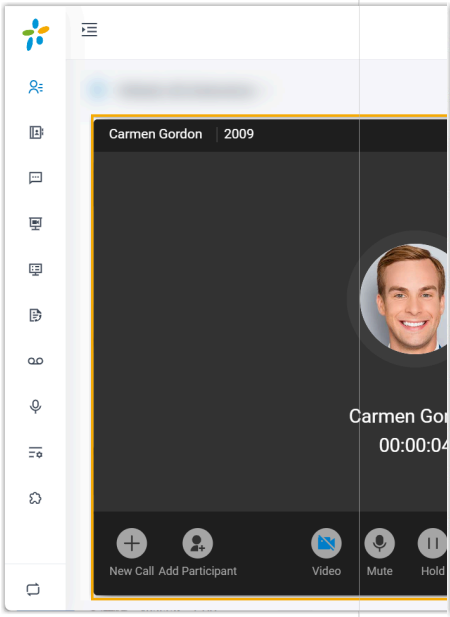
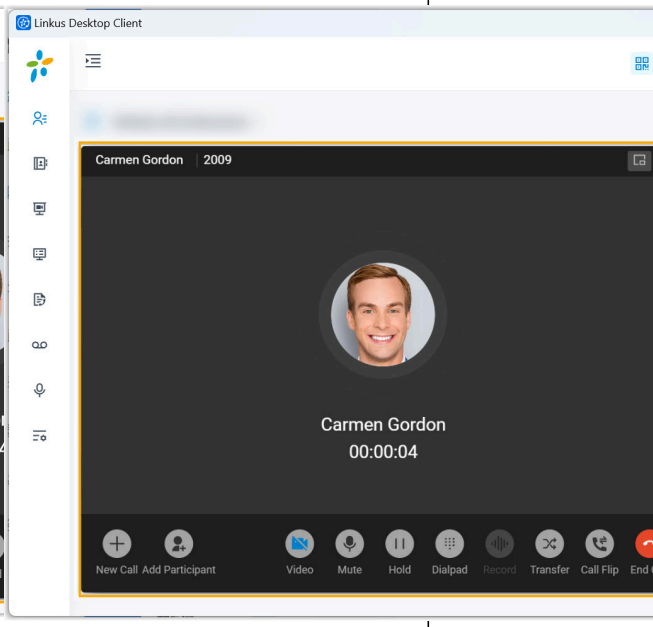
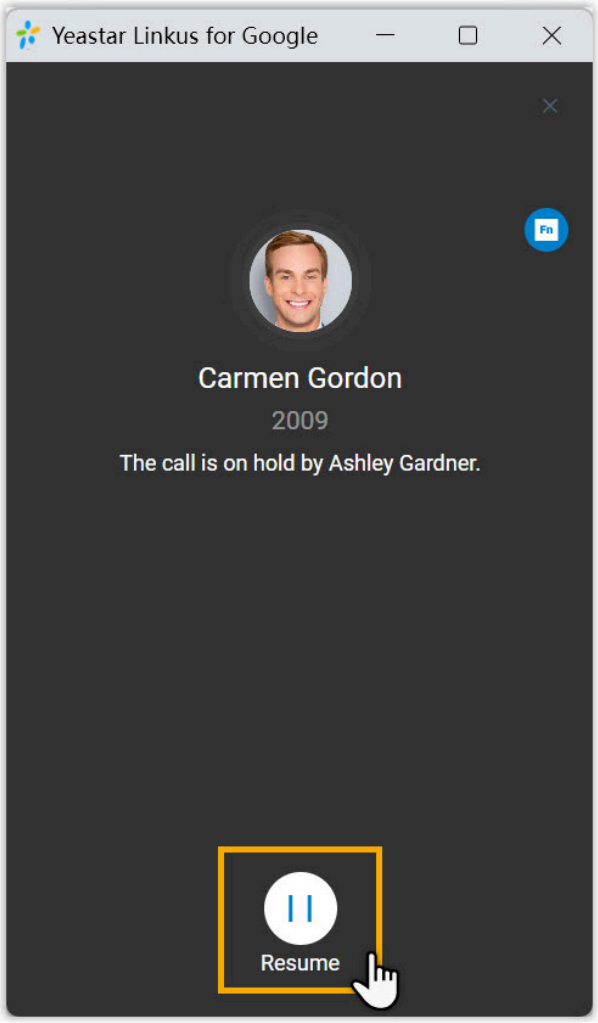
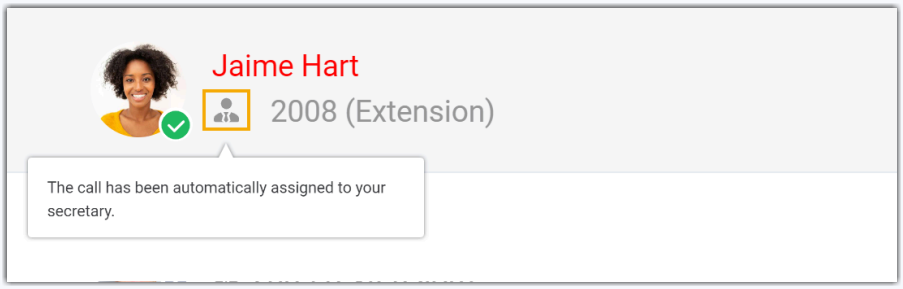
<table>
<tr><th>Endpoint</th><th>Description</th></tr>
<tr><td>Linkus Web Client & Linkus Desktop Client</td><td>Click Resume in the pop-up window to answer the call.
Note:
◦ To use this feature, ensure the Linkus Desktop Client meets the following version requirements:
▪ Windows Desktop: 1.1.2 or later.
▪ Mac Desktop: 1.1.2 or later.
◦ Linkus Mobile Client does NOT support resuming calls transferred from boss/secretary.
Figure 1. Linkus Web Client
 

Figure 2. Linkus Desktop Client
 
</td></tr>
<tr><td>Yeastar Linkus for Google</td><td>Click Resume in 'Yeastar Linkus for Google' to answer the call.</td></tr>
</table>

Endpoint	Description
	
IP phone	Press the BLF key on the IP phone to answer the call. **Important:** To resume the transferred calls via IP phones, you need to set up IP phones for the boss or secretary. ◦ To allow the boss to resume calls transferred from the secretary via an IP phone, see Monitor secretary's call status on an IP phone. ◦ To allow the secretary to resume calls transferred from the boss via an IP phone, see Monitor boss's call status on an IP phone.

Note:

- Boss can disable this feature or change the call type that needs forwarding on their Linkus Web Client or Linkus Desktop Client (path: **Preferences > Features > Boss Extension**), and the updated settings will be synchronized to PBX automatically.
- On the **Call Logs** page of the boss's Linkus clients, missed calls with icon indicate that the calls were automatically forwarded to the secretary extension, distinguishing them from the actual missed calls of the boss extension.



Monitor Call Status

Monitor Secretary's Call Status

With the Boss-Secretary feature enabled, you can set up function keys and configure the IP phone for the boss, allowing the boss to check if the secretary is currently handling his or her call by monitoring the secretary's call status through Linkus clients and IP phone.

Monitor secretary's call status on Linkus clients

By setting up function key(s) for the boss's extension, the boss can monitor secretary's call status through **Linkus Desktop Client**, **Linkus Web Client**, and **Yeastar Linkus for Google**.

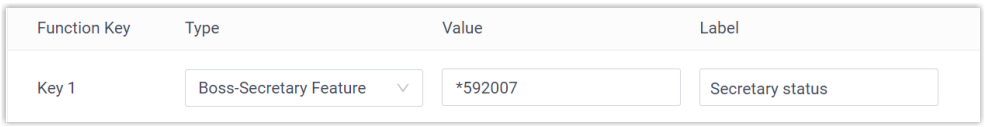
Prerequisites

- You have Set up Boss-Secretary Feature for Extensions.
- Obtain the **Enable Boss-Secretary Feature** feature code.

Note:
The default feature code is `*59`. You can view or update the feature code on **Call Features > Feature Code > Boss-Secretary Feature > Enable Boss-Secretary Feature**.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, select the boss extension and click .
2. Click the **Function Keys** tab.
3. Configure the following settings to subscribe to the secretary's call status.



   - **Type**: Select **Boss-Secretary Feature**.
   - **Value**: Enter the **Enable Boss-Secretary Feature Code** and **Secretary Extension Number**.

     For example, the **Enable Boss-Secretary Feature Code** is `*59`, and the **Secretary Extension Number** is `2007`, then enter `*592007`.
   - **Label**: Optional. Enter a display name for this function key.
4. Click **Save**.

Result

The boss can monitor the secretary's call status by the icons in the **Function Keys** list of Linkus clients.

Icons & Descriptions

- : The secretary is NOT handling any calls for the boss.
- : The secretary is answering calls for the boss.
- : The secretary is putting a call on hold, waiting for the boss to answer.

Linkus clients

Figure 3. **Linkus Web Client & Linkus Desktop Client**

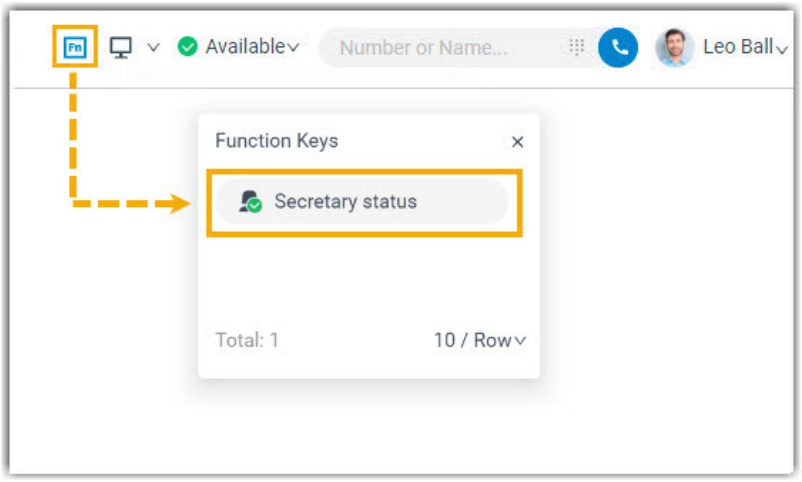
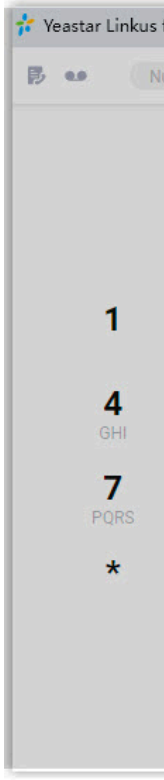


Figure 4. **Yeas**



Monitor secretary's call status on an IP phone

By setting up the boss's IP phone, the boss can monitor the secretary's call status by the BLF LED, and resume calls transferred from the secretary using the BLF key.

Prerequisites

- You have set up function keys for boss to monitor secretary's call status.
- Ensure that the IP phone is connected to Yeastar P-Series Cloud Edition via auto provisioning, and has been assigned to the boss extension.

> **Note:**
> For more information about auto provisioning, see Auto Provision IP Phones.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Click beside the boss's phone.

3. In the pop-up window, click **OK**.

Result

The boss can monitor the secretary's call status through the BLF LED on his or her IP phone:

- **Solid Green**: The secretary is NOT handling any calls for the boss.
- **Solid Red**: The secretary is answering calls for the boss.
- **Flashing Red**: The secretary is putting a call on hold, waiting for the boss to answer. And the boss can press the BLF key to resume and answer the call.

> **Note:**
> If the boss is using a **Fanvil phone**, you need to set the **BLF Hold** LED to either **Slowblink** or **Fastblink**. Otherwise, the BLF LED will not flash when the secretary places a call on hold.

Monitor Boss's Call Status

With the Boss-Secretary feature enabled, you can set up function keys and configure the IP phone for the secretary, allowing the secretary to check if the boss is holding calls for him or her to resume by monitoring the boss's call status on Linkus clients and IP phone.

Monitor boss's call status on Linkus clients

By setting up function key(s) for the secretary's extension, the secretary can monitor the boss's call status through **Linkus Desktop Client**, **Linku Web Client**, and **Yeastar Linkus for Google**.

Prerequisites

- You have Set up Boss-Secretary Feature for Extensions.
- Obtain the **Enable Boss-Secretary Feature** feature code.

> **Note:**
> The default feature code is `*59`. You can view or update the feature code on **Call Features > Feature Code > Boss-Secretary Feature > Enable Boss-Secretary Feature**.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, select the secretary extension and click .
2. Click the **Function Keys** tab.
3. Configure the following settings to subscribe to the boss's call status.



   - **Type**: Select **Boss-Secretary Feature**.
   - **Value**: Enter the **Enable Boss-Secretary Feature Code** and **Boss Extension Number**.

     For example, the **Enable Boss-Secretary Feature Code** is `*59`, and the **Boss Extension Number** is `2010`, then enter `*592010`.
   - **Label**: Optional. Enter a display name for this function key.
4. Click **Save**.

Result

The secretary can check if the boss is currently holding a call by the icons in the **Function Keys** list of Linkus clients.

Icons & Descriptions

- : The boss is NOT holding any calls for the secretary to resume.
- : The boss is putting a call on hold, waiting for the secretary to resume.

Linkus clients

Figure 5. **Linkus Web Client & Linkus Desktop Client**

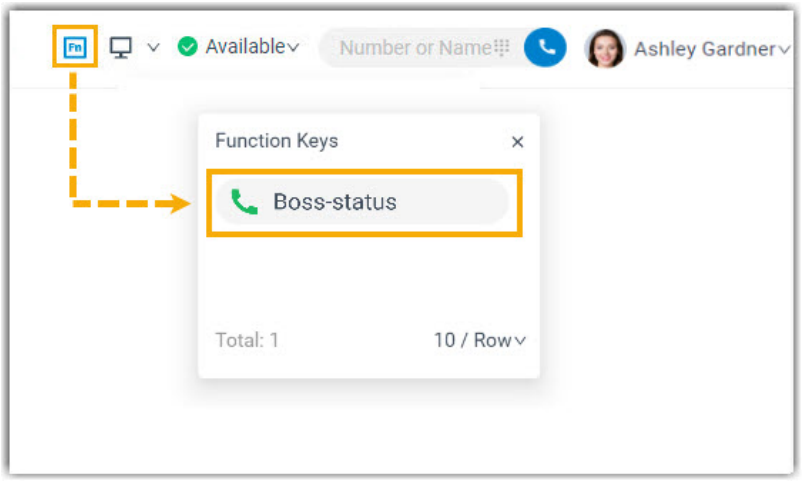
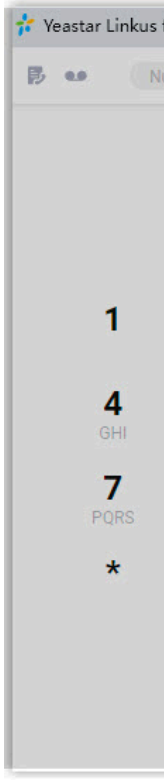


Figure 6. **Yeas**



Monitor boss's call status on an IP phone

By setting up the secretary's IP phone, the secretary can monitor the boss's call status by the BLF LED, and resume calls transferred from the boss using the BLF key.

Prerequisites

- You have set up function keys for secretary to monitor boss's call status.
- Ensure that the IP phone is connected to Yeastar P-Series Cloud Edition via auto provisioning, and has been assigned to the secretary extension.

> **Note:**
> For more information about auto provisioning, see Auto Provision IP Phones.

Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Click ↻ beside the secretary' phone.

3. In the pop-up window, click **OK**.

Result

The secretary can check if the boss is currently holding a call through the BLF LED on his or her IP phone:

- **Solid Green**: The boss is NOT holding any calls for the secretary to resume.
- **Flashing Red**: The boss is putting a call on hold, waiting for the secretary to resume. And the secretary can press the BLF key to resume the call.

> **Note:**
> If the secretary is using a **Fanvil phone**, you need to set the **BLF Hold** LED to either **Slowblink** or **Fastblink**. Otherwise, the BLF LED will not flash when the boss places a call on hold.

Hot Desking

Hot Desking Overview

Hot desking feature allows extension users to log in to a shared desk phone temporarily using their extension number and voicemail PIN, so that users can make and receive calls on the shared phone with their own extension numbers.

Scenario

The Hot Desking feature is ideal for shared offices, field teams, and call centers where employees have flexible schedules or work in shifts.

For example, call centers typically work in shifts to provide a 24-hour service, so employees may need to share a desk phone at different times. With hot desking feature, the employee can log in to a shared phone and answer incoming calls with his or her own extension during working hours. When the employee log out of the shared phone after work, the system automatically logs the user out of the queue, avoiding delays in responding to incoming calls. This not only optimizes the use of telephone hardware resources but also improves the economic benefits of the call center.

Requirements

- **PBX Server**: The firmware of Yeastar P-Series Cloud Edition is 84.10.0.30 or later.
- **IP Phone**: The model and firmware of the IP phone meets the requirements listed in Auto Provisioning - Supported Devices.

Process of setting up hot desking

To implement hot desking, you need to complete the following configurations:

Set up a hot desking phone

You need to set up a hot desking phone via auto provisioning according to your network environment.

For more information, see Set up a Hot Desking Phone.

Enable hot desking for extension users

You need to enable hot desking feature for specific extension users. Only the extension users with hot desking feature enabled can log in to a hot desking phone.

For more information, see Enable Hot Desking for an Extension User.

Methods of using hot desking

After you set up a hot desking phone, authorized users can log in to and log out of the hot desking phone in the following ways:

- Dial a Guest In / Guest Out feature code
- Press a Guest In / Guest Out key

Dial a Guest In / Guest Out feature code

By dialing a feature code on a hot desking phone, users can log in and use the phone with their own extension number, and log out after finish using the phone.

The default hot desking feature code is listed below:

- **Guest In**: *84
- **Guest Out**: *084

Tip:

> You can change, enable, or disable the hot desking feature code on PBX web portal: **Call Features > Feature Code > Hot Desking**.

For more information, see dial a feature code to log in and dial a feature code to log out.

Press a Guest In / Guest Out key

The PBX system automatically assigns a **Guest In** and **Guest Out** key to a hot desking phone for login or logout. Users can press the BLF key on the phone to log in and use the phone with their own extension number, or log out after finish using the phone.

For more information, see press a BLF key to log in and press a BLF key to log out.

Set up Hot Desing

Set up a Hot Desking Phone

This topic describes how to set up a phone for hot desking via auto provisioning.

> **Important:**
> If the desired phone has been assigned to an extension, you need to delete the phone from auto provisioning phone list and RESET the phone first.

Supported devices

Hot desking is applicable on the IP phones listed in Auto Provisioning - Supported Devices.

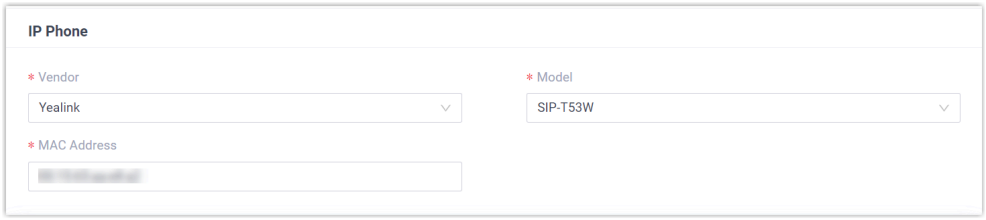
Procedure

Prerequisites

Gather information of IP phone, including Vendor, Model, and MAC address.

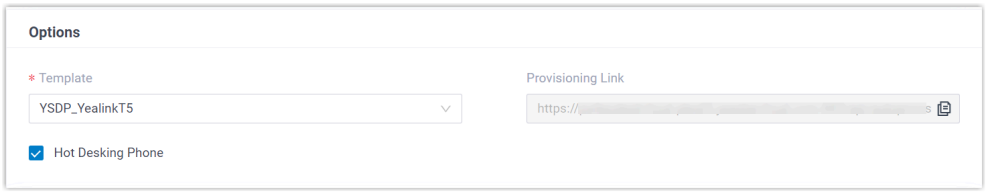
Procedure

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Click **+Add > Add**.
3. In the **IP Phone** section, enter the phone's information:



- **Vendor**: Select a phone vendor.
- **Model**: Select a phone model.
- **MAC Address**: Enter MAC address of the phone.

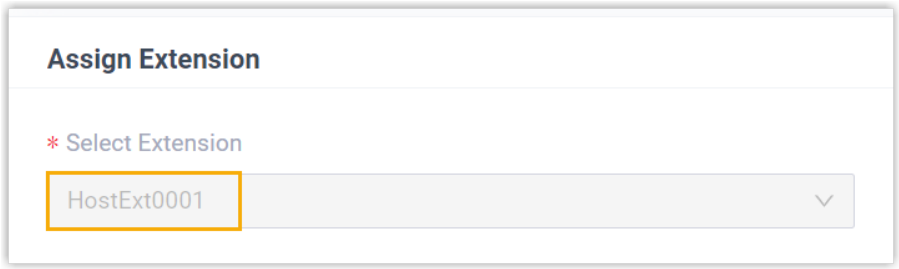
4. In the **Options** section, configure the following settings:



- **Template**: Select a desired template for the phone from the drop-down list.

  A provisioning server URL is generated accordingly and displayed on the **Provisioning Link** field.
- **Hot Desking Phone**: Enable the option to set the phone as a hot desking phone.

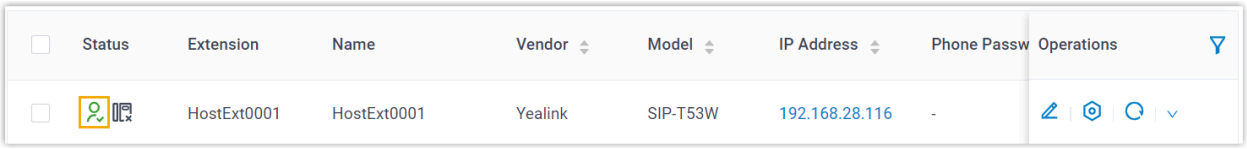
  The phone is automatically assigned to a virtual extension named HostExt*{virtual_num}*.



5. Click **Save**.
6. If the phone you add is Grandstream IP phone or Alcatel-Lucent Enterprise phone, you need to log in to the IP phone web interface and manually add the provisioning link to the phone.
7. Reboot the IP phone manually.

Result

The phone is connected to the PBX and set as a hot desking phone with the virtual extension registered. You can check the registration status from the phone list; The PBX system automatically assigns a **Guest In** key to the phone.



Note:
Users can only use the phone with virtual extension registered to make emergency calls. To use the phone for regular calling, users must either dial the **Guest In** feature code or press the **Guest In** BLF key to log in the phone.

What to do next

Enable Hot Desking for an Extension User

Related information

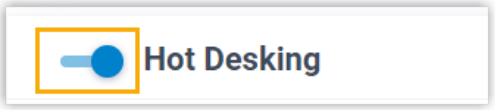
IP Phone Configuration Guide

Enable Hot Desking for an Extension User

This topic describes how to enable hot desking for an extension user, so that the user can log in and use a hot desking phone with his or her own extension number.

Procedure

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click beside the desired extension, then click the **Features** tab.
3. Scroll down to the **Hot Desking** section, then turn on the switch.



4. Configure the following settings according to your needs.

Setting	Description
Log Out of Queue	Decide whether to automatically log out the dynamic agent from a queue. If enabled, the system will automatically log the extension user out of the queue when the user logs out of a hot desking phone.
Automatic Guest Out	Decide whether to automatically log out the extension user from a hot desking phone: • **Never**: Disable automatic logout. • **After**: Specify a time period to log the user out of a hot desking phone since the extension user logs in. • **At Daily**: Specify a fixed time to log the user out of a hot desking phone every day.

5. Click **Save** and **Apply**.

Related information

Log in to a Hot Desking Phone
Log out of a Hot Desking Phone
Manage Hot Desking feature

Use Hot Desking

Log in to a Hot Desking Phone

This topic describes how extension users can log in to a hot desking phone.

Prerequisites

- You have set up a hot desking phone.
- You have enabled hot desking feature for desired extension users.
- If the users have their own phones, and may occasionally log in to a hot desking phone, you need to set concurrent registrations.

Procedure

Extension users can log in to a hot desking phone via either of the following methods:

Method	Instruction
Dial a feature code to log in	1. On a hot desking phone, dial the **Guest In** feature code. **Note:**

Method	Instruction
	The default **Guest In** feature code is *84. You can customize the feature code on **Call Features > Feature Code > Hot Desking > Guest In**. 2. Follow the voice prompt, enter the extension number followed by the # key (e.g. 2000#). 3. Follow the voice prompt, enter the voicemail PIN followed by the # key (e.g. 8471#).
Press a BLF key to log in	After you set up a hot desking phone, the system automatically assigns a **Guest In** key to the phone, extension users can press the key to log in to the phone. 1. On a hot desking phone, press the **Guest In** key. 2. Follow the voice prompt, enter the extension number followed by the # key (e.g. 2000#). 3. Follow the voice prompt, enter the voicemail PIN followed by the # key (e.g. 8471#). **Troubleshooting:** If a user failed to log in to the hot desking phone by pressing the **Guest In** key, you can check if the **Guest In** feature code has been changed (Path: **Call Features > Feature Code > Hot Desking > Guest In**).

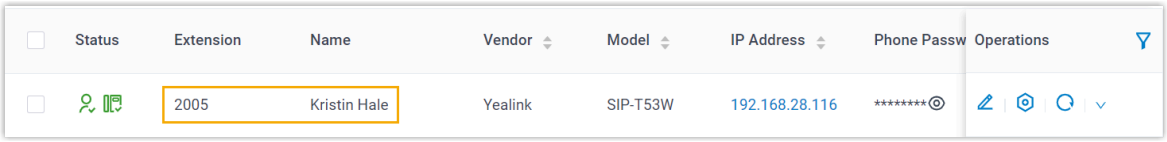
Result

- The user's extension configurations, including extension information and function key settings, are loaded onto the phone; The system automatically assigns a **Guest Out** key to the phone.
- The extension user can use the phone with his or her extension and implement the followings:
  - Make calls from the phone.
  - Receive calls on the phone.
  - Use the function keys of the extension.
  - Query contacts that are stored in the PBX on the IP phone via LDAP.

Note:

> This requires you to configure the LDAP directory settings for the phone on **Auto Provisioning > Phones >** **> LDAP Directory**. For more information about the settings, see Auto Provision LDAP for IP Phones.

- You can monitor the hot desking phone status, and know who is working on the hot desking phone on **Auto Provisioning > Phones**.



Related information

Log out of a Hot Desking Phone

Log out of a Hot Desking Phone

This topic describes how extension users can log out of a hot desking phone.

Procedure

Extension users can log out of a hot desking phone using either of the following methods:

Method	Instruction
Dial a feature code to log out	On a hot desking phone, dial the **Guest Out** feature code. **Note:** The default **Guest Out** feature code is *084. If you want to customize the feature code, go to **Call Features > Feature Code > Hot Desking > Guest Out**.
Press a BLF key to log out	The system automatically assigns a **Guest Out** key to a hot desking phone when a user logged in. The user can press the key on the phone to quickly log out. **Troubleshooting:** If a user failed to log out by pressing the **Guest Out** key, you need to check if the **Guest Out** feature code has been changed (Path: **Call Features > Feature Code > Hot Desking > Guest Out**).

Result

The user's extension is logged out of the phone; the phone is reverted to the default settings and can only be used to make emergency calls.

> **Important:**
> The call logs are kept on the hot desking phone, the user needs to manually clear the data if necessary.

> **Note:**
> If a user forgot to log out, after another user logs in to the hot desking phone, the previous user would be logged out automatically, or you can forcibly log the user out of the phone on PBX web portal.

Related information

Log in to a Hot Desking Phone

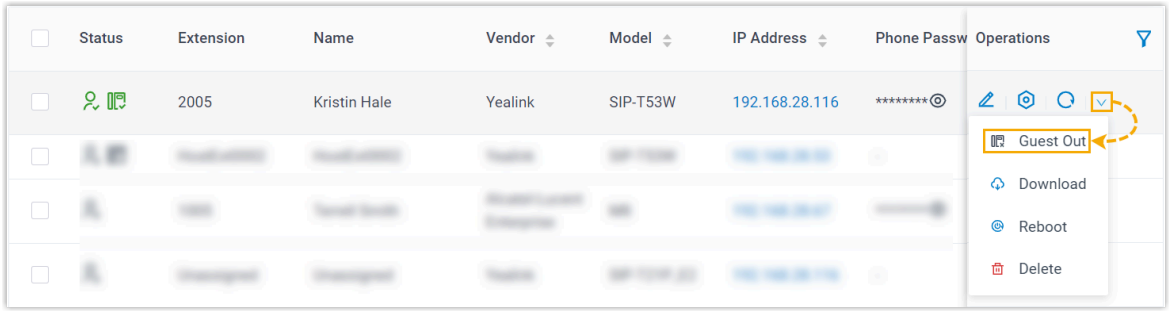
Manage Hot Desking feature

This topic describes how to manage the hot desking feature on PBX web portal, including forcibly log out a user, and disable hot desking for an extension user.

Forcibly log a user out of a hot desking phone

In case an extension user forgot to log out of a hot desking phone, you can forcibly log his or her extension out on PBX web portal, so as to prevent information disclosure or misuse of the extension.

1. Log in to PBX web portal, go to **Auto Provisioning > Phones**.
2. Hover your mouse over beside the desired phone, and click **Guest Out**.

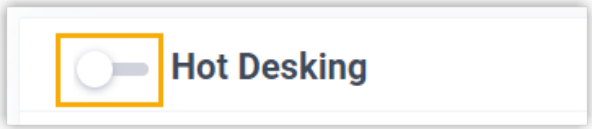


3. In the pop-up window, click **Confirm**.

The user is logged out of the hot desking phone.

Disable Hot Desking for an extension user

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click beside the desired extension, then click the **Features** tab.
3. Scroll down to the **Hot Desking** section, then turn off the switch.



4. Click **Save** and **Apply**.

   The extension user will not be able to log in to a hot desking phone.

Busy Camp-on

Camp on to a Busy Extension

Busy Camp-on feature allows a caller to place a call reservation when he or she makes a call to an extension and receives a busy signal. The PBX will automatically call the caller back as soon as the called extension becomes available.

Restrictions

- The Busy Camp-on feature is only applicable for the calls between extensions.
- The timeout for a Busy Camp-on request is **3600** seconds. This means that the PBX will not call the caller back if the called party is still unavailable 1 hour after the request is initiated.
- A maximum of **10** Busy Camp-on requests is available for each extension. If exceeds, the PBX will cancel the earliest requests.

Procedure

Assume that extension 1000 makes a call to extension 2000, but extension 2000 is in a call.

In this case, extension 1000 can hang up the call and camp on to extension 2000 by dialing a feature code. The PBX will call extension 1000 back and connect it to extension 2000 as soon as extension 2000 becomes available.

1. Extension 1000 dials *792000 to camp on to extension 2000.

   > **Tip:**
   > The default feature code for enabling Busy Camp-on is `*79`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Busy Camp-on > Enable Busy Camp-on**.

   A prompt "The Busy Camp-on has been enabled" is played and the call is hung up automatically.
2. As soon as extension 2000 becomes available, the PBX will call extension 1000.

   Extension 1000 receives an incoming call pop-up with caller ID displayed as **Busy Camp-on** ***{name_of_called_extension}{number_of_called_extension}*** and will hear a prompt "This call is connected via Busy Camp-on feature" after answering the call.
3. The PBX calls extension 2000.

   After extension 2000 answers the call, extension 2000 and extension 1000 are connected.

Cancel Busy Camp-on Requests

If extension users are not available to accept the automatic callback, they can cancel Busy Camp-on requests as shown in the topic.

> **Note:**
> We recommend that extension users think twice before proceeding, as this will cancel all the Busy Camp-on requests that they have initiated.

Procedure

Extension user dials *079 on the phone to cancel all the Busy Camp-on requests that he or she has initiated.

> **Tip:**
> The default feature code for disabling Busy Camp-on is `*079`. You can change, enable, or disable the code on PBX web portal: **Call Features > Feature Code > Busy Camp-on > Disable Busy Camp-on**.

Result

- A prompt "The Busy Camp-on has been disabled" is played and the call is hung up automatically.
- All Busy Camp-on requests from the extension user are cancelled.

Fax

Fax Overview

Yeastar P-Series Cloud Edition allows you to connect your fax machine to PBX system. Then you can send or receive faxes on a fax machine, and receive faxes by email. This topic describes how fax works with Yeastar P-Series Cloud Edition, and introduces fax to email, fax detection, and Fax over VoIP settings.

T.38 Fax

T.38 is a protocol that enables fax over the Internet and is supported on Yeastar P-Series Cloud Edition. T.38 utilizes Voice over IP (VoIP) to send a fax. This process is known as virtual fax or FoIP (Fax over IP).

The diagram below explains how T.38 Fax works:

1. A fax machine sends a fax through a T.38 compatible gateway, which acts as an emitting server.
2. The emitting server partitions data from the fax into an image that can be encoded and sent over the Internet in real time, then sends the T.38 data stream to another T.38 compatible server, such as a PBX, which acts as a receiving server.
3. The receiving server converts the T.38 data stream to analog signal, and sends to the terminal fax machine.



Fax to email

Faxes traditionally are sent directly to a fax machine; the recipient receives a printed copy. Yeastar P-Series Cloud Edition provides fax to email feature that allows you to receive faxes as PDF by email.

The benefits of fax to email:

- Keep your faxes private without paper trail.
- Access faxes in real-time from anywhere.
- No need to pay for expensive hardware, printer paper, ongoing maintenance or a dedicated fax line.

Fax detection

Fax detection is used to detect automatically whether an incoming call is voice or fax. It is useful when you have fax call and voice call on the same line.

- If the PBX detects a fax signal, the PBX immediately routes the call to the designated fax destination.
- If the PBX does not detect a fax signal, the PBX handles the call as a regular voice call.

Fax over IP (FoIP) settings

The following settings are available when you want to improve the Fax transmission over VoIP network.

- **T.38 Support**: Enable or disable T.38 protocol for extension and trunk according to your needs.
- **T.38 Max BitRate**: The maximum bit rate of the fax transmission.

  The default value is **14400**.
- **No T.38 Attributes in re-INVITE SDP**: Whether to contain T.38 attributes in SDP re-invite packet.
- **Error Correction Mode**: Error Correction Mode (ECM) is an optional transmission mode. ECM automatically detects and corrects errors in the fax transmission process that are sometimes caused by telephone line noise.

Receive Faxes by Email

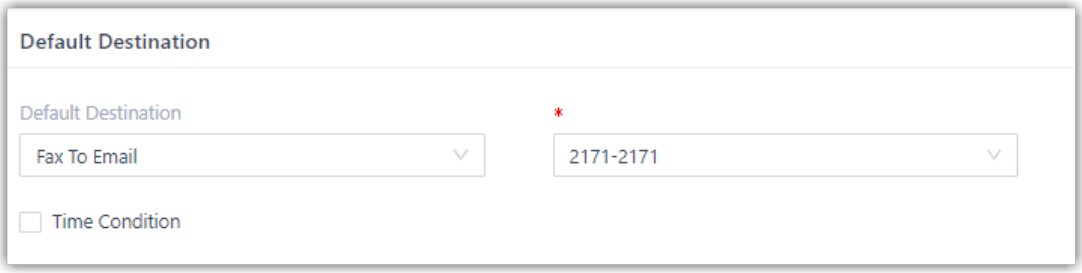
Yeastar P-Series Cloud Edition provides fax to email feature that allows you to receive faxes as PDF by email. This topic describes how to receive faxes by email.

Prerequisites

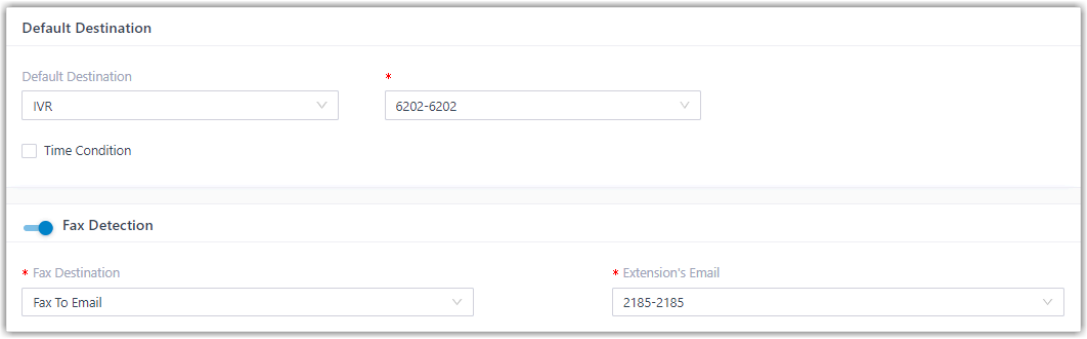
- Make sure the PBX system email works, or the PBX cannot forward the received faxes to an extension user's email.
- Make sure there is a valid email address assigned to extension.
- **Optional**: Customize the fax email template.

Procedure

1. Log in to PBX web portal, go to **Call Control > Inbound Route**, edit the inbound route for incoming faxes.
2. If you receive faxes through a dedicated line, go to **Default Destination** section.
   a. In the **Default Destination** drop-down list, select **Fax To Email**.
   b. Select an extension user to receive faxes by email.



3. If you receive faxes through a shared line, go to **Fax Detection** section.
   a. In the **Fax Destination** drop-down list, select **Fax To Email**.
   b. In the **Extension's Email** drop-down list, select an extension user to receive faxes by email.



4. Click **Save** and **Apply**.

Result

When receiving a fax, PBX converts the received fax and simply forwards it to the email address as an PDF attachment.

Set up Fax over IP (FoIP)

Fax over IP (FoIP) is the process of using T.38 protocol to send a fax from a fax machine to another fax machine over the Internet. This topic describes how to enable T.38 for extension and trunk respectively, and how to change T.38 settings to improve the Fax transmission over VoIP network.

Enable T.38 protocol for SIP extension

If you want to register a SIP extension on a SIP compatible fax machine, you need to enable **T.38 Support** for this extension.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**, edit the desired extension.
2. Go to **Advanced > VoIP Settings**.
3. Select the checkbox of **T.38 Support**.
4. Click **Save** and **Apply**.

Enable T.38 protocol for SIP trunk

If you want to use a SIP trunk to send or receive faxes, you need to enable **T.38 Support** for this trunk.

1. Log in to PBX web portal, go to **Extension and Trunk > Trunk**, edit the desired trunk.
2. Go to **Advanced > VoIP Settings**.
3. Select the checkbox of **T.38 Support**.
4. Click **Save** and **Apply**.

Change T.38 settings

If the Fax over IP doesn't work, you can change the T.38 settings.

1. Log in to PBX web portal, go to **PBX Settings > SIP Settings > T.38**.
2. Change the T.38 settings.

- **T.38 Max BitRate**: Set the maximum bit rate of the fax transmission.
- **No T.38 Attributes in re-INVITE SDP**: If enabled, SDP re-invite packet does not contain T.38 attributes.
- **Error Correction Mode**: If enabled, after receiving the packet for a complete fax page, PBX notifies the transmitting fax machine of the frames with errors. The transmitting fax machine then retransmits the specified frames.

  This process is repeated until all frames are received without errors.

3. Click **Save** and **Apply**.

Paging/Intercom

Overview of Paging and Intercom

This topic describes what is Paging and Intercom, scheduled paging call and intercom call.

What is Paging and Intercom

Yeastar P-Series Cloud Edition Paging and Intercom feature helps users broadcast announcements over one or more speakers, without the called party picking up the handset.

Paging

Paging feature is used to make a one-way announcement to users via a phone speaker.

When a broadcaster makes a paging call, the group members' phones automatically answer into speakerphone mode. Group members can not talk with the broadcaster during the call.

For more information, see Set up a One-way Paging Group.

Intercom

Intercom feature is used to establish two-way communication with users via a phone speaker.

When a broadcaster makes an intercom call, the group members' phones automatically answer into speakerphone mode. The broadcaster and all the group members can talk with each other during the call.

For more information, see Set up a Two-way Intercom Group.

Scheduled paging call and intercom call

Besides real-time paging calls or intercom calls, you can set a time schedule to automatically start your broadcast. The Scheduled Paging/Intercom feature is perfect for schools, airports, or other facilities that require routine notifications set in advance.

For more information, see Schedule a Paging Call or an Intercom Call.

Paging/Intercom Group

Set up a One-way Paging Group

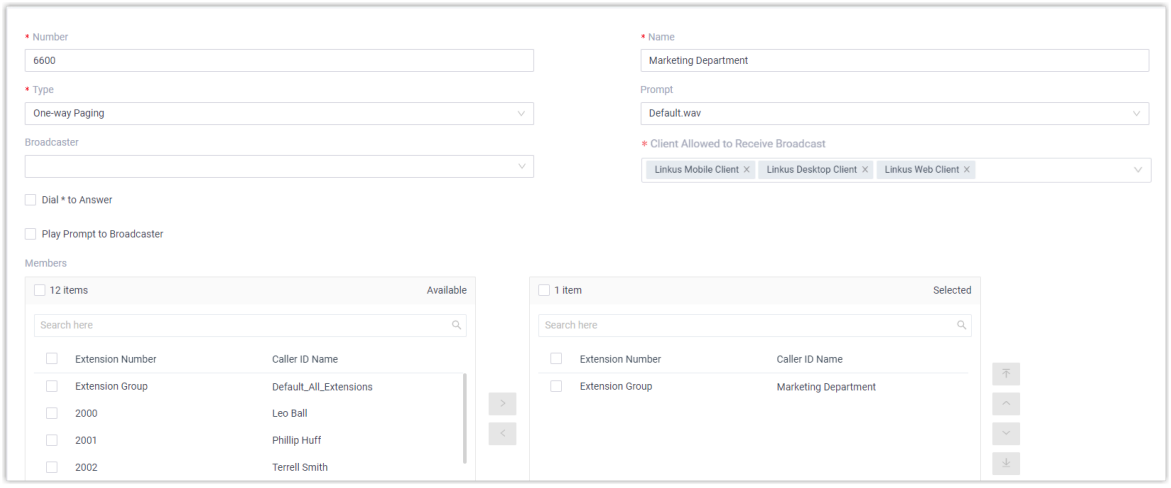
One-way Paging feature allows a broadcaster to make an announcement to users. The called parties' phones will not ring, but instead directly answering into speakerphone mode. This topic describes how to set up a one-way paging group.

Scenario

A company has different departments on different floors in a building. Each department is deployed with a phone for communication. The boss has an urgent case that needs to confirm with marketers. In this case, you can set up a One-way paging group for Marketing Department. The boss can make a paging call to the department and ask marketers concerned to go to the office.

Procedure

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**, click **Add**.
2. Configure a one-way paging group.



- **Number**: Enter a number for the paging group. In this example, enter *6600*.
- **Name**: Enter a name for the paging group. In this example, enter *Marketing Department*.
- **Type**: Select **One-way Paging**.
- **Prompt**: Optional. To play a prompt before making an announcement, you can select a custom prompt. In this example, select **Default.wav**.

> **Note:**
> To customize a prompt, see Record a Custom Prompt or Upload a Custom Prompt.

- **Broadcaster**: Optional. To restrict users from making an announcement to the paging group, select allowed extensions or extension groups from the drop-down list. In this example, leave it blank.
- **Client Allowed to Receive Broadcast**: Select the desired clients or extension endpoint for receiving broadcasts.
- **Dial * to Answer**: Optional. To allow users to dial `*` to talk to the broadcaster privately, enable this option. In this example, keep the option disabled.

> **Note:**
> When a user dials `*`, announcement will terminate, and the user can have a private talk with the broadcaster.

- **Play Prompt to Broadcaster**: Optional. To restrict the broadcaster from hearing the prompt, disable this option. In this example, disable this option.

> **Note:**
> This option is available only when a custom prompt is selected and is enabled by default.

- **Members**: Select desired members from **Available** box to **Selected** box. In this example, select **Marketing Department**.

3. Click **Save** and **Apply**.

What to do next

When dialing the number of the paging group, the selected members will receive broadcast through the client(s) you have specified.

- For Linkus clients (Mobile/Web/Desktop/Pad), they will ring first, and users need to manually answer the call to listen to the broadcasts.

> **Note:**
> Members can set their **Linkus Mobile Client** to automatically answer paging calls (Path: **Account > Settings > Advanced > Auto Answer Paging/Intercom**).

- For IP phones, they will automatically answer into speakerphone mode and play the broadcasts.

Related information

Schedule a Paging Call or an Intercom Call

Set up a Two-way Intercom Group

Two-way Intercom feature allows you to establish two-way communication with an individual user or a group of users. The called parties can respond without picking up the handset. This topic describes how to set up a two-way intercom group.

Background information

In office complexes, hospitals, or schools, there are either static guards or patrol guards to ensure safety within the workplace. The Two-way Intercom feature helps improve communication efficiency. For example, a security guard can ask for help when security incidents happen, a supervisor can flexibly dispatch employees in daily activities.

Yeastar P-Series Cloud Edition supports to place an intercom call to one or more users:

Place an intercom call to a specific user

Dial Intercom feature code (default: *6) followed by a desired extension number.

For example, dial *61002 to place an intercom call to 1002.

> **Tip:**
> To change intercom feature code, go to **Call Features > Feature Code > Intercom**.

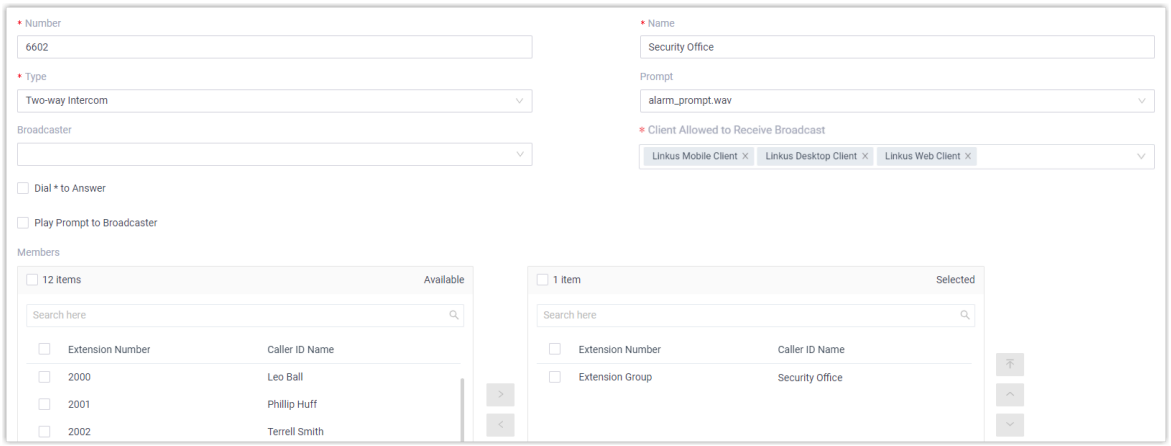
Place an intercom call to multiple users

Set up a two-way intercom group on the PBX and place a call to group numbers.

For more information, see the following instructions.

Procedure

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**, click **Add**.
2. Configure a two-way intercom group.



- **Number**: Enter a number for the intercom group. In this example, enter *6602*.
- **Name**: Enter a name for the intercom group. In this example, enter *Security Office*.
- **Type**: Select **Two-way Intercom**.
- **Prompt**: Optional. To play a prompt before making an announcement, you can select a custom prompt. In this example, select **alarm_prompt.wav**.

> **Note:**
> To customize a prompt, see Record a Custom Prompt or Upload a Custom Prompt.

- **Broadcaster**: Optional. To restrict users from placing an intercom call to the intercom group, select allowed extensions or extension groups from the drop-down list. In this example, leave it blank.
- **Client Allowed to Receive Broadcast**: Select the desired clients or extension endpoint for receiving intercom calls.

> **Note:**

> Intercom calls initiated by the feature code will NOT be restricted by this configuration and will call all the clients and extension endpoints of members simultaneously.

- **Dial * to Answer**: Optional. To allow users to dial `*` to talk to the broadcaster privately, enable this option.

> **Note:**
> When a user dials `*`, the call is ended from other users' side, and the user can have a private talk with the broadcaster.

- **Play Prompt to Broadcaster**: Optional. To restrict the broadcaster from hearing the prompt, disable this option. In this example, disable this option.

> **Note:**
> This option is available only when a custom prompt is selected and is enabled by default.

- **Members**: Select desired members from **Available** box to **Selected** box. In this example, select the group *Security Office*.

3. Click **Save** and **Apply**.

What to do next

When dialing the number of the intercom group, the selected members will receive the call through the client(s) you have specified.

- For Linkus clients (Mobile/Web/Desktop/Pad), they will ring first, and users need to manually answer the call.

> **Note:**
> Members can set their **Linkus Mobile Client** to automatically answer intercom calls (Path: **Account > Settings > Advanced > Auto Answer Paging/Intercom**).

- For IP phones, they will automatically answer into speakerphone mode.

Related information

Schedule a Paging Call or an Intercom Call

Manage Paging Groups and Intercom Groups

This topic describes how to edit or delete paging groups and intercom groups.

Edit a paging/intercom group

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**.
2. On **Paging/Intercom List** page, click beside desired group.
3. Edit group settings.
4. Click **Save** and **Apply**.

Delete a paging/intercom group

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**.
2. On **Paging/Intercom List** page, click beside desired group.
3. Click **OK** and **Apply**.

> **Note:**
> If you have scheduled a paging call or an intercom call for the group, the scheduled call will also be deleted.

Scheduled Paging/Intercom Call

Schedule a Paging Call or an Intercom Call

A scheduled paging call or intercom call allows Yeastar P-Series Cloud Edition or an extension user to make an announcement at a specific date and time. For facilities that require routine notifications set in advance, you can schedule a paging call or an intercom call.

Prerequisites

You have set up a paging group or an intercom group.

- Set up a One-way Paging Group
- Set up a Two-way Intercom Group

Procedure

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**, click **Scheduled Paging/Intercom** tab.
2. Schedule a paging call or an intercom call.
   a. Click **Add**.
   b. Configure the following settings:
      - **Paging**: Select a pre-configured paging group from the drop-down list.
      - **Caller**: Select a broadcaster.
        - *{extension_user}*: The extension user will make the announcement.
          On the specified date and time, the PBX will place a call to the user. When the user answers the call, group members' phones directly answer into speakerphone mode.

          > **Note:**
          > If the user rejects the call, the announcement will be cancelled.

        - **None**: The PBX will make the announcement.
          On the specified date and time, the PBX will place a call to group members and play a specific custom prompt. After the prompt ends, the PBX hangs up. The option can be applied to school bells, church bells, etc.

          > **Note:**
          > The option is available only when a custom prompt is assigned to the selected paging group or intercom group.

      - **Start Date**: Set the start date of the scheduled paging call or intercom call.
      - **Time**: Set the start time of the scheduled paging call or intercom call.

        > **Note:**
        > You can set up to 8 timings, which means that the paging call or intercom call can be placed at different time on the same day.

      - **Days of Week**: Select the days of week.

The scheduled paging call or intercom call will be weekly placed on the specified days of week.

c. Click **Save** and **Apply**.

Manage Scheduled Paging Calls and Intercom Calls

This topic describes how to edit or delete scheduled paging calls and intercom calls.

Edit a scheduled paging/intercom call

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**.
2. On **Scheduled Paging/Intercom** page, click  beside desired group.
3. Edit relevant settings.
4. Click **Save** and **Apply**.

Delete a scheduled paging/intercom call

1. Log in to PBX web portal, go to **Call Features > Paging/Intercom**.
2. On **Scheduled Paging/Intercom** page, click  beside desired group.
3. Click **OK** and **Apply**.

   The announcement will not be made on the specified date and time.

PIN List

Add a PIN List

A PIN list allows you to define groups and then assign a list of passwords to each group. The PIN list can be used to restrict outbound routes to enhance communication security. Users need to enter a correct PIN code when making outbound calls through a restricted outbound route.

Procedure

1. Log in to PBX web portal, go to **Call Features > PIN List**, click **Add**.
2. In the pop-up window, configure the following settings:
   - **Name**: Specify a name to help you identify it.

- **PIN List**: Enter the PIN codes. Press the **Enter** key to separate multiple PIN codes.

  > **Note:**
  > - The PIN code only allows numeric value.
  > - The length of each PIN code is limited from 3 to 15.

- **Record in CDR**: Whether to record the PIN code in CDR when the PIN code has been used.

3. Click **Save**.

What to do next

1. Assign the PIN codes included in the PIN list to different users.
2. Select a PIN list in an outbound route to restrict outbound calls.

Blocked/Allowed Numbers

Block Calls To or From a Phone Number

Yeastar P-Series Cloud Edition supports to block incoming and/or outgoing calls by phone number. To stop nuisance calls, you can add phone numbers to the system blocklist. Numbers in the blocklist will be blocked to dial in, dial out, or both.

Restriction

- Blocked numbers do NOT work for the extensions within the PBX. When an extension number matches a blocked number, the extension can still be used for outgoing and incoming calls.
- The maximum number of **Blocked Numbers Lists** and **Numbers per Blocked Numbers List** varies depending on the number of your extensions.

Table 28.

Maximum Number of Extensions (N)	Blocked Numbers Lists	Numbers per Blocked Numbers List
N ≤ 200	256	100
N >200	512	200

Background information

Yeastar P-Series Cloud Edition allows you to handle calls by phone number in the following ways:

Call Handling Based on Caller ID

This feature makes it possible for routing or blocking incoming calls from internal or external users by phone number. You can customize different call handling rules for each extension.

For more information, see Handle Incoming Calls Based on Caller ID.

Blocked Numbers

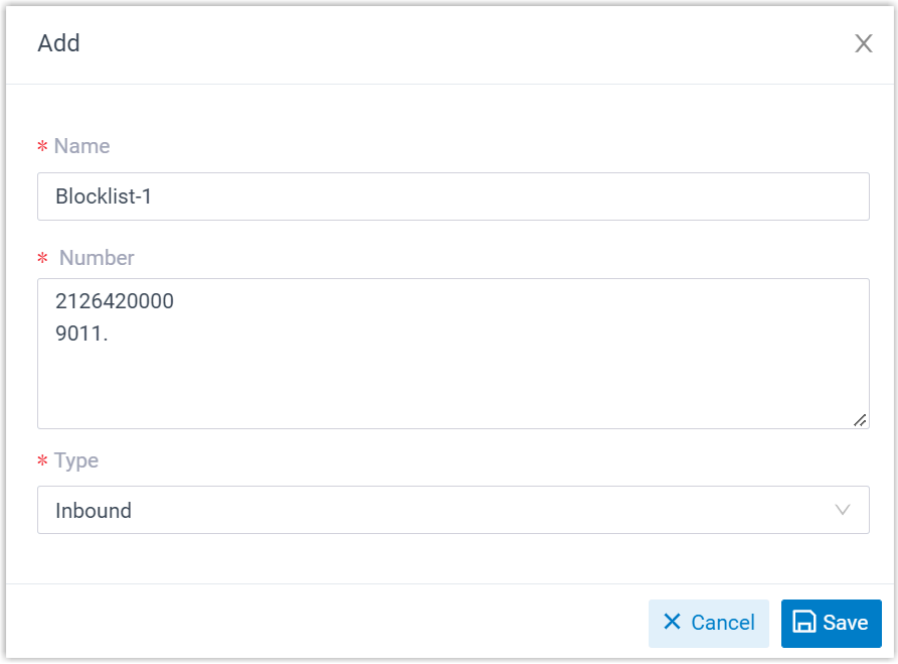
This feature makes it possible for blocking inbound and/or outbound calls to external users by phone number. If you list a phone number in the system blocklist, all the PBX extensions can NOT reach or be reached by the phone number.

For more information, see the following instructions.

> **Note:**
> System blocklist has higher priority than individual blocklist.

Procedure

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Blocked Numbers**.
2. Click **Add** to set up a blocked number list.
3. In the pop-up window, configure as follows:



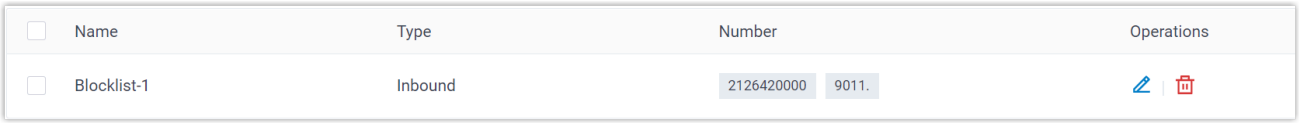
- **Name**: Enter a name to help you identify the number(s) to be blocked.
- **Number**: Enter a specific number or a number pattern per line.
  - To block a specific number, enter a specific number. For example, enter `2126420000`.
  - To block a range of numbers, enter a wildcard pattern. For example. enter `9011.` to block numbers starting with 9011.

    For more information about wildcard pattern, see DID Pattern and Caller ID Pattern.
- **Type**: Select a type from the drop-down list.
  - **Inbound**: Block the number(s) from calling into the PBX.
  - **Outbound**: Block PBX extensions from calling the number(s).
  - **Both**: Block the number(s) from calling into the PBX and block the PBX extensions from calling the number(s).

4. Click **Save** and **Apply**.

Result

The blocked numbers list is displayed on the web page as the following figure shows. The added numbers will be blocked based on the type you selected.



Export and Import Blocked Numbers

The blocked numbers added on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired blocked numbers information in the exported file, and import the file to PBX again. This topic describes how to export and import blocked numbers.

Export all blocked numbers

You can export all the blocked numbers to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Blocked Numbers**.
2. Click **Export**.

A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Blocked Numbers Parameters.

Import blocked numbers

We recommend that you export blocked numbers to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file

- **Format**: UTF-8 .csv
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Blocked Numbers Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Blocked Numbers**.
2. Click **Import**.

3. In the pop-up window, click **Browse** to select the UTF-8.csv file you prepared.
4. Click **Import**.

Result

The blocked numbers in the CSV file are displayed in the **Blocked Numbers** list.

Related information

Import and Export -FAQ

Manage Blocked Numbers

This topic describes how to edit and delete blocked numbers lists.

Edit blocked numbers lists

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Blocked Numbers**.
2. Click beside a desired list.
3. In the pop-up window, edit the name, the blocked number(s), or blocked type as needed.
4. Click **Save** and **Apply**.

Delete blocked numbers lists

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Blocked Numbers**.
2. To delete a blocked numbers list, do as follows:
   a. Select the checkbox of a desired list, then click .
   b. In the pop-up window, click **OK**.
3. To bulk delete blocked numbers lists, do as follows:
   a. Select the checkboxes of desired lists, then click **Delete**.
   b. In the pop-up window, click **OK**.

The blocked numbers lists are deleted successfully. All the numbers in the deleted lists are no longer blocked, they can call into the PBX and be called by PBX extension users.

Allow Calls To or From a Phone Number

If trusted phone numbers happen to be listed in system blocklist, you can add the trusted phone numbers to system allowlist. Numbers in the allowlist are allowed to dial in, dial out, or both.

> **Note:**
> The **Allowed Number** has higher priority than the **Blocked Number**; Adding numbers in **Allowed Number** doesn't mean that PBX only allow these numbers to dial in or be dialed out.

Limitations

The maximum number of **Allowed Numbers Lists** and **Numbers per Allowed Numbers List** varies depending on the number of your extensions.

Table 29.

Maximum Number of Extensions (N)	Allowed Numbers Lists	Numbers per Allowed Numbers List
N ≤ 200	256	100
N >200	512	200

Background information

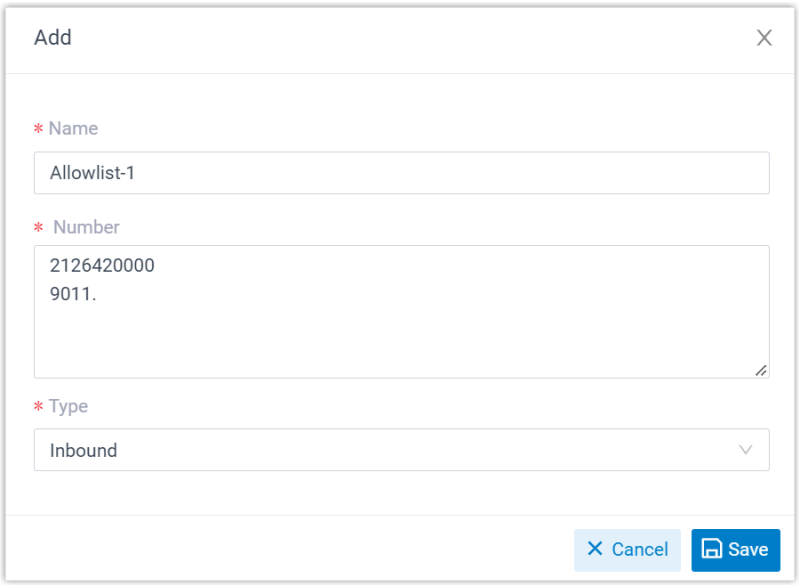
If your customers' phone numbers happen to be listed in system blocklist or individual blocklist, you can add trusted phone numbers to the allowlist.

To add trusted phone numbers to individual allowlist, see Handle Incoming Calls Based on Caller ID.

To add trusted phone numbers to system allowlist, see the following instructions.

Procedure

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Allowed Numbers** .
2. Click **Add** to set up an allowed number list.
3. In the pop-up window, configure as follows:



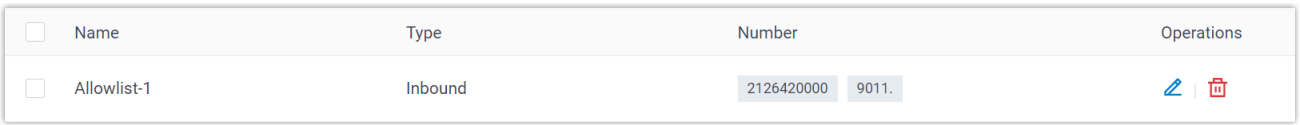
- **Name**: Enter a name to help you identify the number(s) to be allowed.
- **Number**: Enter a specific number or a number pattern per line.
  - To allow a specific number, enter a specific number. For example, enter `2126420000`.
  - To allow a range of numbers, enter a wildcard pattern. For example. enter `9011.` to allow numbers starting with 9011.

    For more information about wildcard pattern, see DID Pattern and Caller ID Pattern.
- **Type**: Select a type from the drop-down list.
  - **Inbound**: Allow the number(s) to call into the PBX.
  - **Outbound**: Allow PBX extensions to call the number(s).
  - **Both**: Allow the number(s) to call into the PBX and allow PBX extensions to call the number(s).

4. Click **Save** and **Apply**.

Result

The allowed numbers list is displayed on the web page as the following figure shows. The added numbers can communicate with the PBX extensions based on the type you selected.



Export and Import Allowed Numbers

The allowed numbers added on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired allowed numbers information in the exported file, and import the file to PBX again. This topic describes how to export and import allowed numbers.

Export all allowed numbers

You can export all the allowed numbers to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Allowed Numbers** .
2. Click **Export**.

A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Allowed Numbers Parameters.

Import allowed numbers

We recommend that you export allowed numbers to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8 .csv
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information, see Allowed Numbers Parameters.

Procedure

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Allowed Numbers** .
2. Click **Import**.
3. In the pop-up window, click **Browse** to select the UTF-8.csv file you prepared.
4. Click **Import**.

Result

The allowed numbers in the CSV file are displayed in the **Allowed Numbers** list.

Related information

Import and Export -FAQ

Manage Allowed Numbers

This topic describes how to edit and delete allowed numbers lists.

Edit allowed numbers lists

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Allowed Numbers** .
2. Click beside a desired list.
3. In the pop-up window, edit the name, the allowed number(s), or type as needed.
4. Click **Save** and **Apply**.

Delete allowed numbers lists

1. Log in to PBX web portal, go to **Call Features > Blocked/Allowed Numbers > Allowed Numbers** .
2. To delete an allowed numbers list, do as follows:
   a. Select the checkbox of a desired list, then click .
   b. In the pop-up window, click **OK**.
3. To bulk delete allowed numbers lists, do as follows:
   a. Select the checkboxes of desired lists, then click **Delete**.
   b. In the pop-up window, click **OK**.

The allowed numbers lists are deleted successfully. If the numbers in the deleted lists match the numbers or the number patterns from Blocked Numbers, they would be blocked based on the blocking type.

Messaging

Yeastar P-Series Cloud Edition Omnichannel Messaging Overview

Yeastar P-Series Cloud Edition offers omnichannel messaging feature, which allows a business to integrate different messaging channels with the business phone system. In this way, customers are able to reach the business through SMS and social media they prefer, while business agents can centrally receive and respond to customers' queries sent from different messaging channels on their Linkus UC Clients.

Requirements

- **Firmware**: Version 84.12.0.32 or later
- **Plan**: Enterprise Plan (EP) or Ultimate Plan (UP)

Feature highlights

All-in-one Message Inbox

Agents are able to receive and respond to customers' queries from different messaging channels directly on their Linkus UC Clients, greatly saving their time by eliminating the need of switching between apps or services to check for messages. The messages are stored on the PBX server, providing a central record of all the messaging sessions.

Customer Contact using Business Number

Agents can contact customers using a business number, while keeping their own personal mobile number private. If necessary, the messaging session can be easily elevated to a call to reach the customer, so that the agent can resolve issues faster via voice call.

Seamless Collaboration across Agents

Agents can hand off customers' issue to another agent by transferring the conversation, the new agent can quickly review the whole chat history and take over the conversation without hassle.

Automatic Chat Assignment

Route your business messages from different messaging channels to agents, who can share the workload across teams to reduce customer service response time. The system automatically assigns chats to the first agent that picks up the session from the queue.

Configuration guide

For more detailed information and configurations of Omnichannel Messaging, see Omnichannel Messaging Administrator Guide.

PBX System

System Preferences

This topic describes the preference settings that will be applied globally to Yeastar P-Series Cloud Edition.

Go to **PBX Settings > Preferences** to configure preferences settings.

Basic preferences

Table 30.

Setting	Description
Device Name	Set a name for the PBX. The name will be used as the sender name when PBX sends emails out.
Name Display Format	Set display format for extension user's name and contact's name. • **First Name Last Name with Space Inbetween** • **Last Name First Name with Space Inbetween** • **Last Name First Name without Space Inbetween**
Max Call Duration (s)	Set the global maximum call duration for an active call. When the call duration reaches the limit, the call will be ended. The default value is **10800**. **Note:** For outbound calls, the **Max Call Duration (s)** setting of the caller's extension takes precedence.
Tone Region	Select your country or the nearest neighboring country to enable the default dial tone, busy tone, and ring tone.

Organization Management

Table 31.

Setting	Description
Organization Management	If enabled, you can arrange extension users into organizations.
Company Name	Set your company name, which will be used as the root organization name.

Distinctive Caller ID Name

Table 32.

Setting	Description
Display Call Feature Name	If enabled, the Caller ID will display the originated name when users receive a call from a ring group, queue, and IVR.
Display DID/DDI Name	If enabled, the Caller ID will display the DID name of the source trunk.

DTMF preferences

Setting	Description
DTMF Passthrough	If enabled, PBX will pass DTMF tones directly to the other end without processing the DTMF tones.
DTMF Duration (ms)	Set the duration (in millisecond) of DTMF audio signal sent by the PBX. The default value is **120**.
DTMF Gap (ms)	Set the interval (in millisecond) between two DTMF audio signals sent by the PBX. The default value is **120**.

Extension preferences

Below are default extension ranges. You can change the extension range according to your needs.

Note:
PBX treats the followings as extensions. Extension users can dial extension numbers to reach them directly.

Extension Type	Default Range
User Extension	1000 - 5999
Parking Extension	6000 - 6099
Group Voicemail Extension	6100 - 6199
IVR Extension	6200 - 6299
Ring Group Extension	6300 - 6399
Queue Extension	6400 - 6499
Conference Extension	6500 - 6599
Paging Extension	6600 - 6699

Extension Type	Default Range
Account Trunk	6700 - 6799
WebRTC Trunk Number	6800 - 6899

Voice Prompt

Voice Prompt Overview

This topic describes the definition, types, and preference settings of voice prompt on Yeastar P-Series Cloud Edition.

What is a voice prompt

A voice prompt is a recorded audio message that is played to callers. The voice prompt can be a request that requires callers to input data through DTMF, or an intermediary that provides instructions and directions to help callers obtain information.

Voice prompt types

Yeastar P-Series Cloud Edition supports 3 types of voice prompt:

- **System Prompt**: System prompt is Yeastar-provided prompt to provide instructions for callers. For example, if a password is required for a meeting, users will be prompted to enter password before they successfully join the meeting.

  You can use pre-defined system prompt, or change system prompt by downloading online prompts or uploading custom system prompts.

  For more information, see Change System Prompt and Customize System Prompt.
- **Custom Prompt**: Custom prompt can be company-specific prompt, which is used in specific call scenario. For example, when a call is forwarded to another destination, the caller will be prompted that the call is forwarded.

  You can record new prompt on your phone, or upload pre-recorded prompt to the PBX.

  For more information, see Record a Custom Prompt and Upload a Custom Prompt.
- **Music on Hold**: Music on Hold (MoH) is the business practice of playing recorded music to fill the silence that would be heard by callers who have been placed on hold.

  You can use pre-defined music on hold, or customize your own music on hold.

  For more information, see Set up a Custom MoH Playlist.

Voice prompt preference settings

Navigation path: **PBX Settings > Voice Prompt > Prompt Preferences**.

Table 33.

Setting	Description
Music on Hold	The playlist to be played when a call is on hold. **Note:** The available playlists are synchronized with playlists in **Music on Hold**.
Music on Hold for Call Forwarding	The music to be played when the caller is put on hold during call forwarding. • **Music on Hold**: Play **Music on Hold** to the caller. • **Ringing Tone**: Play ringing tone to the caller.
Invalid Phone Number Prompt	The prompt to be played when a callee number is invalid. **Note:** The available prompts are synchronized with **Custom Prompt**.
Busy Line Prompt	The prompt to be played when a trunk is in use. **Note:** The available prompts are synchronized with **Custom Prompt**.
Call Failure Prompt	The prompt to be played when a call is failed to be sent out. **Note:** The available prompts are synchronized with **Custom Prompt**.
Event Notification Prompt	The prompt to be played when PBX places a call to notify callee that a specific event occurs. **Note:** The available prompts are synchronized with **Custom Prompt**.

Table 33. (continued)

Setting	Description
Play Call Forwarding Prompt	Whether to inform the user that the current call will be forwarded.
Play Call Waiting Prompt	The prompt to be played when the caller is waiting to be connected to an extension user that is on a call. • If the caller is an **extension user**, the system will play call waiting prompt "Sorry! Please hold on, the number you dialed is busy now.", then play ringing tone. • If the caller is an **external user**, the system will play call waiting prompt "Sorry! Please hold on, the number you dialed is busy now.", then play ringing tone or ringback tone (depending on whether you have configured ringback tone for the inbound route from which the caller is routed: **Call Control > Inbound Route > Default Destination > Ringback Tone**).

System Prompt

Change System Prompt

This topic describes how to download an online prompt and change it to the default system prompt.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > System Prompt**.
2. Download the desired system prompt.
   a. Click **Download Online Prompts**.

      All the supported system prompts are displayed on **Download Online Prompts** page.

   b. Select a prompt, click .

   c. Click to close the window.

      The downloaded prompt is displayed on **System Prompt** list.
3. In the **Default** column, set the desired system prompt to default.
4. Click **Save** and **Apply**.

Result

The prompt is applied to the system.

Customize System Prompt

This topic describes how to customize system prompt and change it to the default prompt.

Background information

Yeastar P-Series Cloud Edition provides multiple online prompts for your choice. If you want to use custom system prompt, you need to contact Yeastar Support to record your own prompt, and upload it to your PBX.

Prerequisites

- Contact Yeastar Support to record your own prompt.
- The prompt file must meet the following requirements:
  - **File Type**: `.tar`
  - **File Name**: Special characters are NOT allowed.
  - **File Size**: Up to 100 MB

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > System Prompt**.
2. Upload the custom system prompt.
   a. Click **Upload System Prompts**.
   b. In the pop-up window, select a `.tar` file from your local PC, click **Open**.

   The uploaded prompt file is displayed on **System Prompt** list.
3. In the **Default** column, set the desired system prompt to default.
4. Click **Save** and **Apply**.

Result

The prompt is applied to the system.

Music on Hold

Set up a Custom MoH Playlist

Music on Hold (MoH) is intended to reassure callers that they are connected to their calls. Yeastar P-Series Cloud Edition has a default local audio playlist with built-in MoH files. This topic describes how to set up and use a custom MoH playlist.

Background information

Yeastar P-Series Cloud Edition supports two types of MoH playlists:

- Local audio MoH playlist
- Streaming music MoH playlist

Playlist Type	Description
Local audio MoH playlist	This type of playlist contains a list of audio files that are uploaded to the system, and play back when a caller is placed on hold. For more information, see Set up a local audio MoH playlist.
Streaming music MoH playlist	This type of playlist contains a URL that is used to connect to a live audio feed from a particular source. For more information, see Set up a streaming music MoH playlist.

Set up a local audio MoH playlist

Requirements

The audio files to be uploaded must meet the following requirements:

Item	Requirements
File Format	`.wav`, `.mp3`, or `.gsm` • PCM, 8K, 16bit, 128kbps • A-law(g.711), 8k, 8bit, 64kbps • u-law(g.711), 8k, 8bit, 64kbps **Tip:** If file format does not meet the requirement, you can convert audio files via WavePad or G711 File Converter online.

Item	Requirements
File Size	Up to 8 MB

Limitations

Item	Limitations
Max. local audio MoH playlists	16
Max. audio files in a playlist	8

Step1. Add a local audio MoH playlist

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Music on Hold**.
2. Create a new playlist.
   a. Click **Create New Playlist**.
   b. In the pop-up window, configure the playlist.
      - **Playlist Type**: Select **Local Audio**.
      - **Playlist Name**: Enter a name to help you identify it.
      - **Play Order**: Decide whether to play the playlist alphabetically or randomly.
   c. Click **Save**.
3. Add one or more audio files to the playlist.
   a. Select the created playlist, click  .
   b. In the pop-up window, click **Upload**.
   c. Click **Browse** to choose the desired audio file, then click **Upload**.
   d. **Optional:** To add more audio files, repeat **step b-c**.

   The uploaded audio files are displayed on the **MoH Files** list.
4. **Optional:** Check sound quality and completeness of the audio files.
   a. On **MoH Files** page, select the desired audio file, click  .
   b. In the pop-up window, set where to play the audio file.
      - In the **Play on Web** section, click  to play the audio file.
      - In the **Extension** drop-down list, select an extension and click **Play**.

      PBX will call and play the audio file to the extension.
   c. Click **OK**.

5. Click **Apply**.

Step2. Change the system MoH playlist

1. Click **Prompt Preferences** tab.
2. In the **Music on Hold** drop-down list, select the desired playlist.
3. Click **Save** and **Apply**.

Result

When a call is put on hold, the system will play audio files in the playlist to the waiting party.

Set up a streaming music MoH playlist

Limitation

Max. streaming music MoH playlists: 5

Step1. Add a streaming music MoH playlist

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Music on Hold**.
2. Create a new playlist.
    a. Click **Create New Playlist**.
    b. In the pop-up window, configure the playlist.
        - **Playlist Type**: Select **Streaming Music**.
        - **Playlist Name**: Enter a name to help you identify it.
        - **Streaming Music URL**: Enter the URL of an existing music stream.
    c. Click **Save**.
3. Click **Apply**.

Step2. Change the system MoH playlist

1. Click **Prompt Preferences** tab.
2. In the **Music on Hold** drop-down list, select the desired playlist.
3. Click **Save** and **Apply**.

Result

When a call is put on hold, the system will play the streaming music from the URL to the waiting party.

> **Note:**
> If PBX can not access the URL or if there is no available audio in the URL, the system will not play any sound to the waiting party.

Manage MoH Playlists

This topic describes how to edit or delete MoH playlists.

Edit a MoH playlist

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Music on Hold**.
2. Select the desired playlist, click .
3. Edit the playlist according to your needs.
   - **Playlist Name**: Change the playlist name.
   - **Play Order**: Decide whether to play the playlist alphabetically or randomly.

     > **Note:**
     > This option is only available for **Local Audio MoH Playlists**.

   - **Streaming Music URL**: Change the URL of the streaming music.

     > **Note:**
     > This option is only available for **Streaming Music MoH Playlists**.

4. Click **Save** and **Apply**.

Delete MoH playlists

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Music on Hold**.
2. Select the desired playlist, click .
3. In the pop-up dialog box, click **OK**.
4. Click **Apply**.

If the deleted playlist is used for **Music on Hold** or **Music on Hold for Call Forwarding**, the system will not play any sound to the party who is put on hold during a call or call forwarding.

Manage MoH Audio Files

This topic describes how to manage audio files of a local audio MoH playlist.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Music on Hold**.
2. In the **Operations** column, click MoH beside the desired local audio MoH playlist.
3. In the pop-up window, manage MoH audio files according to your needs, then click **OK**.
   - To upload an audio file, click **Upload** and select the desired file.
   - To listen to an audio file, click , decide whether to play the audio file to an extension or on web.
   - To download an audio file, click .
   - To delete an audio file, click .
4. Click **OK** and **Apply**.

Configure Call Forwarding Prompt

This topic describes how to configure call forwarding prompt.

Background information

Call forwarding prompt is used to prompt a caller that the call is forwarded to another destination. By default, when PBX is forwarding an incoming call to another number, the PBX will play the call forwarding prompt "please hold when I try to locate the person you are calling", and then play the MoH music. If you do not want the caller to find out that the call is being forwarded, you can disable **Play Call Forwarding Prompt**.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Prompt Preferences**.
2. Unselect the checkbox of **Play Call Forwarding Prompt**.

3. **Optional:** To change MoH music, select **Music on Hold** or **Ringing Tone** from the drop-down list of **Music on Hold for Call Forwarding**.

   > **Note:**
   > The **Music on Hold** is the playlist that you have defined in **Music on Hold** (**PBX Settings > Voice Prompt > Prompt Preferences > Music on Hold**).

4. Click **Save** and **Apply**.

Custom Prompt

Record a Custom Prompt

This topic describes how to record a custom prompt on a phone.

Prerequisites

At least one extension is ready for use.

Limitation

Maximum Number of Extensions (N)	N ≤50	50 < N ≤ 200	N > 200
Max number of Custom Prompts	64	128	128

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**.
2. Record a custom prompt.
   a. Click **Record New**.

      A window pops up.
   b. In the **Name** field, enter a name to help you identify the prompt.
   c. In the **Extension** drop-down list, select an extension to record the prompt.
   d. Click **Record**.

      The system places a call to the selected extension. After you answer the call, you will hear a prompt for the recording.
   e. Record your prompt on the phone.

When done, hang up or press the `#` key.

Result

Refresh the web page and click **Custom Prompt** tab.

The recorded prompt is displayed on the **Custom Prompt** page.

- To listen to the prompt, click ▶.
- To change the voice content, click 🎙 to record again.

Upload a Custom Prompt

This topic describes how to upload a custom prompt.

Prerequisites

Prepare an audio file that meets the following requirements:

- **File format**: `.wav`, `.mp3`, or `.gsm`
  - PCM, 8K, 16bit, 128kbps
  - A-law(g.711), 8k, 8bit, 64kbps
  - u-law(g.711), 8k, 8bit, 64kbps

> **Tip:**
> If the audio file does not meet the requirements, you can convert the audio file via WavePad or G711 File Converter online.

- **File size**: Up to 8MB.

Limitation

Maximum Number of Extensions (N)	N ≤50	50 < N ≤ 200	N > 200
Max number of Custom Prompts	64	128	128

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**.

2. Click **Upload**.
3. In the pop-up window, select an audio file from your local PC and click **Open**.

Result

The uploaded file is displayed on **Custom Prompt** page.

Manage Custom Prompts

This topic describes how to manage custom prompts, such as re-record, play, download, and delete a prompt.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Voice Prompt > Custom Prompt**.
2. In the **Operations** column, manage custom prompts according to your needs.
    - To re-record a prompt, click , select an extension to record.
    - To listen to a prompt, click , decide whether to play the audio file to an extension or on web.
    - To download a prompt, click .
    - To delete a prompt, click , click **OK** and **Apply**.

Convert Audio Files

This topic describes how to convert audio files via WavePad or G711 File Converter online.

Background information

Audio files to be uploaded as MoH files or custom prompts must meet the requirements. If your audio file does not meet the requirement, you can use audio editor to convert file format.

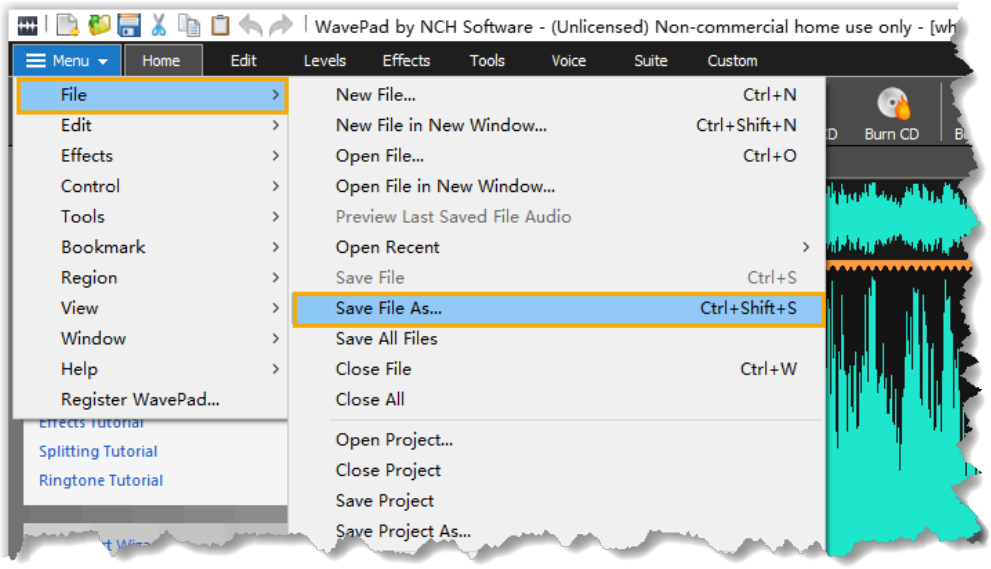
In this topic, we take the followings as examples to show you how to convert file format.

- Convert Audio Files via WavePad
- Convert Audio Files Online

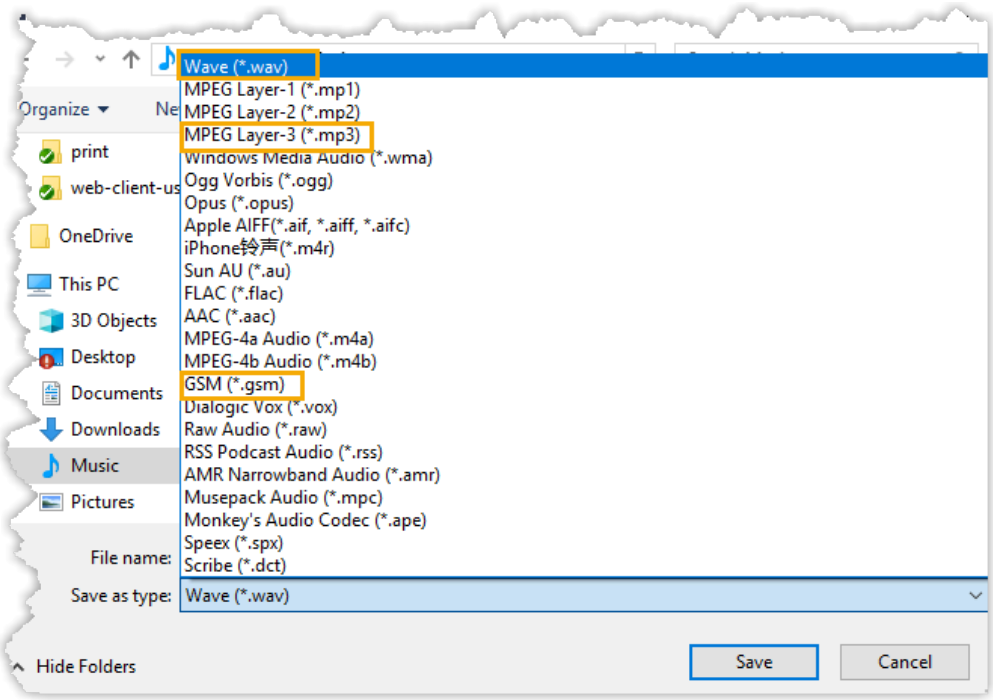
Convert audio files via WavePad

To use WavePad to convert audio files to new formats, download WavePad to your local PC, and proceed as follows.

1. Launch WavePad, open your audio file.
2. Click **File > Save File As**.

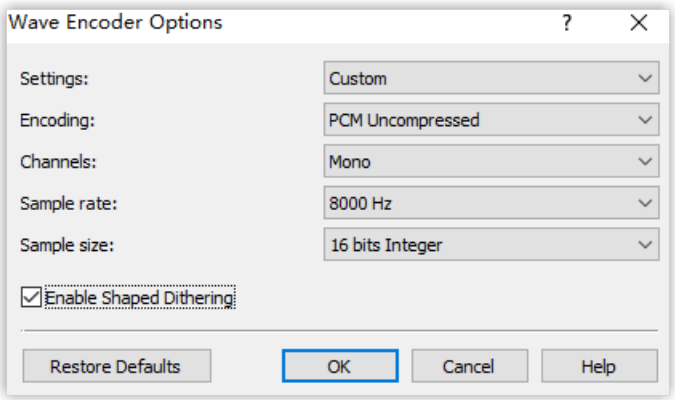


3. In the **Save as type** drop-down list, select `Wave(*.wav)`, `MPEG Layer-3(*.mp3)`, or `GSM(*.gsm)`, click **Save**.



4. If you save the audio file as `Wave(*.wav)` or `MPEG Layer-3(*.mp3)`, you need to configure the encoder options and click **OK**.

> **Note:**
> Select any one of the encoders, and configure relevant options as below.
> - PCM Uncompressed, Mono, 8000 Hz, 16 bits Integer
> - CCITT A-law, Mono, 8000 Hz, 8 bits Integer
> - CCITT u-law, Mono, 8000 Hz, 8 bits Integer

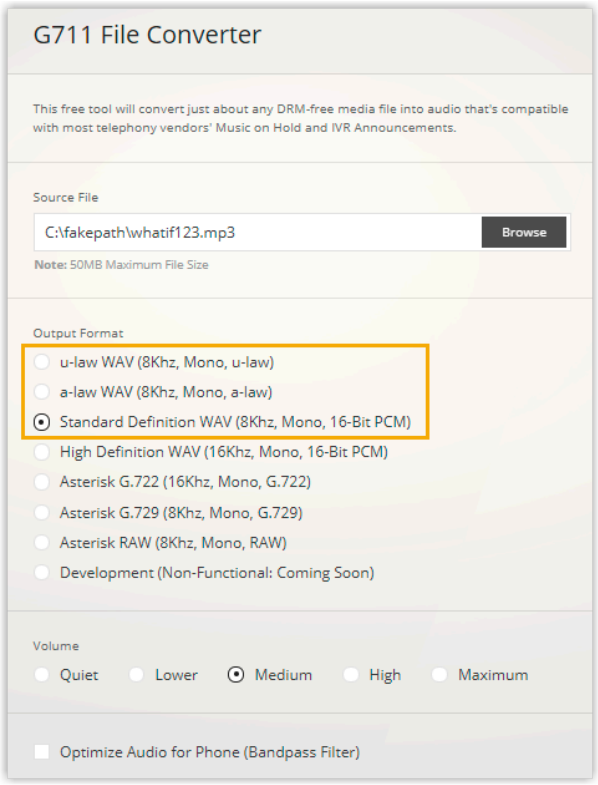


Convert audio files online

If you don't want to download an app, you can quickly convert your audio files via G711 File Converter online.

1. Go to g711.org.
2. Click **Browse** to upload your audio file.
3. Set the **Output Format**.

> **Note:**
> Select any one of **u-law WAV (8Khz, Mono, u-law)**, **a-law WAV (8Khz, Mono, a-law)**, and **Standard Definition WAV (8Khz, Mono, 16-Bit PCM)**.



4. Click **Submit** to start converting the file.

Audio Files Requirements

This topic describes the requirements for audio files to be uploaded to Yeastar P-Series Cloud Edition.

Applications of audio files

You may need to upload a custom audio file in the following scenarios:

- Voicemail greetings
- Custom prompt
- Local audio MoH playlists

Audio file requirements

Audio files to be uploaded to the PBX must meet the following requirements:

Option	Requirement
File Name	Should NOT contain special characters.
File Size	Up to 8 MB.
File Format	`.wav`, `.mp3`, or `.gsm`. • PCM, 8Khz, 16bit, 128kbps • A-law (g.711), 8Khz, 8bit, 64kbps • u-law (g.711), 8Khz, 8bit, 64kbps

SIP Settings

This topic describes the SIP settings on the Yeastar P-Series Cloud Edition for reference.

The SIP configurations require professional knowledge of SIP protocol, incorrect configuration may cause calling issues on the SIP extensions and SIP trunks.

Go to **PBX Settings > SIP Settings** to configure SIP settings.

SIP general settings

Table 34.

Setting	Description
Basic Settings	
SIP UDP Port	UDP Port used for SIP registration. The default value is 5060.
SIP TCP Port	TCP Port used for SIP registration. The default value is 5060.
SIP Endpoint Registration Timer	

Table 34. (continued)

Setting	Description
Max Registration Time (s)	Maximum duration (in seconds) of incoming registrations and subscriptions.
Min Registration Time (s)	Minimum duration (in seconds) of incoming registrations and subscriptions.
Qualify Frequency (s)	How often to send SIP OPTIONS packet to SIP device to check if the device is up.
Outbound SIP Registration Timer	
Registration Attempts	The number of registration attempts before giving up (**0** indicates no limit).
Default Registration Time(s)	Default registration duration (in seconds). **Note:** The actual duration needs to subtract 10 seconds from the value you fill in.
SIP Endpoint Subscription Timer	
Max Subscription Time(s)	Maximum duration (in seconds) of incoming subscriptions.
Min Subscription Time(s)	Minimum duration (in seconds) of incoming subscriptions.

SIP codec

A codec is a compression or decompression algorithm used in the transmission of voice packets over a network or the Internet.

Table 35.

Setting	Description
iLBC Mode	The iLBC codec supports the following modes: • **20 ms** • **30 ms** To get better voice quality, you need to set the iLBC mode according to your SIP endpoints.
Codec Selection	Select the codec. **Available values**: u-law, a-law, GSM, H264, VP8, H263, H263P, iLBC, G722, G726, SPEEX, ADPCM, G729A, MPEG4, Opus.

Table 35. (continued)

Setting	Description
	Note: • To ensure that users can have audio calls on Linkus Web Client, you must enable at least any one of **u-law**, **a-law**, or **G722**. • To ensure that users can have video calls on Linkus Web Client after you subscribe Yeastar P-Series Ultimate Plan, you must enable either **VP8** or **H264**.We recommend that you enable **VP8** or set **VP8** to a higher priority.

TLS settings

Setting	Description
TLS	Enable or disable TLS.
SIP TLS Port	TLS port used for SIP registration. The default value is **5061**.
When PBX acting as a Client	
TLS Connection Method	Specify a protocol for outbound client connections. • TLS V1.0 • TLS V1.2 **Note:** It's recommended to use the more secure TLS V1.2.

Session Timer

A periodic refreshing of a SIP session that allows both user agent and proxy to determine if the SIP session is still active.

Setting	Description
Session Timer	Select a session timer mode. • **No**: Do not include “timer” value in any field. • **Supported**: Include “timer” value in Supported header. • **Required**: Include “timer” value in Required header. • **Forced**: Include “timer” value in both Supported and Required header.

Setting	Description
Session-Expires (s)	The max refresh interval in seconds.
Min-SE (s)	The min refresh interval in seconds. The value should not be smaller than 90.

QoS

Quality of Service (QoS) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due to interference from other traffic of lower priority.

When the network capacity is insufficient, QoS can provide users with priority by setting the value.

Setting	Description
ToS (Type of Service)	
ToS SIP	Type of Service for SIP packets.
ToS Audio	Type of Service for RTP audio packets.
ToS Video	Type of Service for RTP video packets.
CoS (Class of Service)	
Cos SIP	Class of Service for SIP packets.
Cos Audio	Class of Service for RTP audio packets.
Cos Video	Class of Service for RTP video packets.

T.38

Adjust T.38 settings if T.38 Fax doesn't work.

Setting	Description
T.38 Max BitRate	Adjust the max BitRate for T.38 fax.
No T.38 Attributes in re-INVITE SDP	If enabled, SDP re-invite packet will not contain T.38 attributes.
Error Correction Mode	Enable or disable Error Correction for the fax.

Advanced SIP settings

Setting	Description
Incoming Caller ID/DID Retrieval	
Get Caller ID From	Decide the system will retrieve Caller ID from which header field.

Setting	Description
	• From • Contact • Remote-Party-ID • P-Asserted-Identity • P-Preferred-Identity
Get DID From	Decide the system will retrieve DID from which header field. • To • Invite • Diversion • Remote-Party-ID • P-Asserted-Identity • P-Preferred-Identity • P-Called-Party-ID **Note:** If **Remote-Party-ID** is selected but the SIP trunk doesn't support this, the system will retrieve DID from **Invite** header.
SIP Request Header	
User Agent	Set the user agent that will be included when sending SIP packages out.
Internal Alert Info	Set an "alert info text" to add to Alert-info header in INVITE request for internal calls. When receiving an internal call, the phone will inspect "Alert-Info" header to determine which ring tone it should use for ringing.
Other Options	
Support Message Request	Whether to support SIP Message Request or not.
Inband Progress	Whether to enable inband progress or not. The Inband Progress setting applies to all the extensions. **Note:** To configure global Inband Progress setting, you need to contact Yeastar support to configure a custom configuration file. • **Check this option**: PBX will send a 183 Session Progress to the extension when told to indicate ringing and immediately start sending ringing as audio.

Setting	Description
	• **Uncheck this option**: PBX will send a 180 Ringing to the extension when told to indicate ringing, but will NOT send it as audio.
Enable uaCSTA Connection	If this option is enabled, the PBX will allow user agent Computer Supported Telecommunications Application (uaCSTA) to remotely control the IP phone via Linkus Web Client CTI or Linkus Desktop Client CTI. **Note:** Your IP phone should support uaCSTA standard to use this function.

Jitter Buffer

Jitter Buffer Overview

This topic describes what is and when to use jitter buffer, and introduces two jitter buffer types supported on Yeastar P-Series Cloud Edition.

What is jitter buffer

Jitter is a variation between the time that voice packets are sent and received. For example, two packets may arrive at the same time, or out of order due to network congestion, which can cause the problem of audio quality. In this case, jitter buffer can be used to arrange packets according to their expected timing values.

Jitter buffer types

Yeastar P-Series Cloud Edition supports two types of jitter buffer:

- **Fixed jitter buffer**: The fixed jitter buffer has a fixed size and the packets leaving the jitter buffer have a constant delay.
- **Adaptive jitter buffer**: Adapting to network's delay, the adaptive jitter buffer has a variable size and the packets leaving the jitter buffer have a variable delay.

When to use jitter buffer

If you have networking issues like packet loss or packets arriving out of order, you can enable jitter buffer to improve call quality.

Packets loss

If the packets are partially lost, the jitter buffer inserts the lost frame and passes them on in an evenly spaced continuous stream.

Packets arriving out of order

If the arriving packets are out of order, the jitter buffer inserts the packets into the buffer in the correct order, and passes them on in the expected order.

For more information of jitter buffer configuration, see Configure Jitter Buffer.

Configure Jitter Buffer

This topic describes how to configure jitter buffer on Yeastar P-Series Cloud Edition.

Background information

If you have networking issues like packet loss or packets arriving out of order, you can enable jitter buffer to improve call quality.

Procedure

1. Log in to PBX web portal, go to **PBX Settings > Jitter Buffer**.
2. Enable **Jitter Buffer**.
3. To enable jitter buffer for trunks, select the desired trunks from **Available** box to **Selected** box.

   The outbound audio through the selected trunk will be dejittered on the other side.
4. To enable jitter buffer for extensions, select the desired extensions from **Available** box to **Selected** box.

   The received audio on the selected extensions will be dejittered.
5. In the **Implementation** drop-down list, select the implementation of jitter buffer.
   - **Adaptive**: Adapting to network's delay, the adaptive jitter buffer has a variable size and the packets leaving the jitter buffer have a variable delay. If you choose the option, specify the adjustment size and the max jitter buffer size as follows.
     - **Adaptive Adjustment Size (ms)**: The size of each adaptive adjustment of jitter buffer. The default value is 50. If you retain the default value, the jitter buffer size will be adjusted dynamically based on current network condition. It will start from 0 ms and grow at a size of 50 ms each time.
     - **Max Jitter Buffer Size (ms)**: The maximum value of adaptive jitter buffer. The default value is 200.

- **Fixed**: The fixed jitter buffer has a fixed size and the packets leaving the jitter buffer have a constant delay.

  If you choose the option, enter a value in the **Jitter Buffer Size (ms)** field. The default value is 200.

6. Click **Save** and **Apply**.

Date and Time

Change System Time Manually

In case you want to change system time when the PBX can not access the Internet, you can change system time manually. This topic describes how to manually change system time to your local time.

Background information

To ensure that the time of logs and CDRs generated on Yeastar P-Series Cloud Edition is consistent with your local time, you need to adjust system time to your local time.

Procedure

1. Log in to PBX web portal, go to **System > Date and Time**.
2. In the **Date and Time** section, set your local date and time.
   a. In the **Time Zone** drop-down list, select your current time zone.
   b. **Optional:** Configure **Daylight Saving Time** according to your needs.
3. In the **Display Format** section, set the display format of date and time.
   - **Date Display Format**
     - **Year/Month/Day**
     - **Month/Day/Year**
     - **Day/Month/Year**
   - **Time Display Format**
     - **12-hour format**
     - **24-hour format**
4. Click **Save** and **Apply**.
5. Reboot the PBX to take effect.

Result

The current system time is updated; the time of logs and CDRs are also updated.

Email Server

Email Server Overview

This topic describes SMTP server, email template, email daily sending limit, and email sent logs.

Email server

Emails to users or the administrator are required in the following situations:

- Send Linkus welcome email.
- Send fax to email.
- Send voicemail to email.
- Send event notifications.

You can use the default SMTP server or custom SMTP server to send emails.

For the default SMTP server, see Set up Default SMTP Server as an Email Server.

For the custom SMTP server, see Set up Gmail as an Email Server and Set up Outlook as an Email Server.

Email template

Yeastar P-Series Cloud Edition has default email templates for different events, you can also customize email templates according to your needs.

For more information, see Customize Email Templates.

Email daily sending limit

If you use custom email server to send emails, you need to know that email server may limit the number of emails that users can send per day to keep system healthy and account safe.

Yeastar P-Series Cloud Edition obtains the quantity from the email server. If reaching the sending limit, users can NOT send emails via the email server.

Email sent logs

Yeastar P-Series Cloud Edition provides email sent logs, which allows you to monitor mail delivery, and offers you error messages to help you troubleshoot delivery issues more quickly.

For more information, see Email Sent Logs.

Set up Default SMTP Server as an Email Server

This topic describes how to set up default SMTP server as the email server of Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **System > Email > Email Server**.
2. In the **Type of Email Server** drop-down list, select **Built-in SMTP Server**.
3. Test if the email server can successfully send emails.
    a. Click **Test**.
    b. In the pop-up window, enter a recipient's email address in the **Email Address** field.
    c. Click **Test**.

Result

- If the test email is sent successfully, the page displays "Success" and the recipient's mailbox would receive the email.
- If the test email is failed to be sent, the page displays "Failed to send" and prompts you an error message. You can check the error in Email Sent Logs.

Set up Gmail as an Email Server

This topic describes how to set up Gmail as an email server in Yeastar P-Series Cloud Edition.

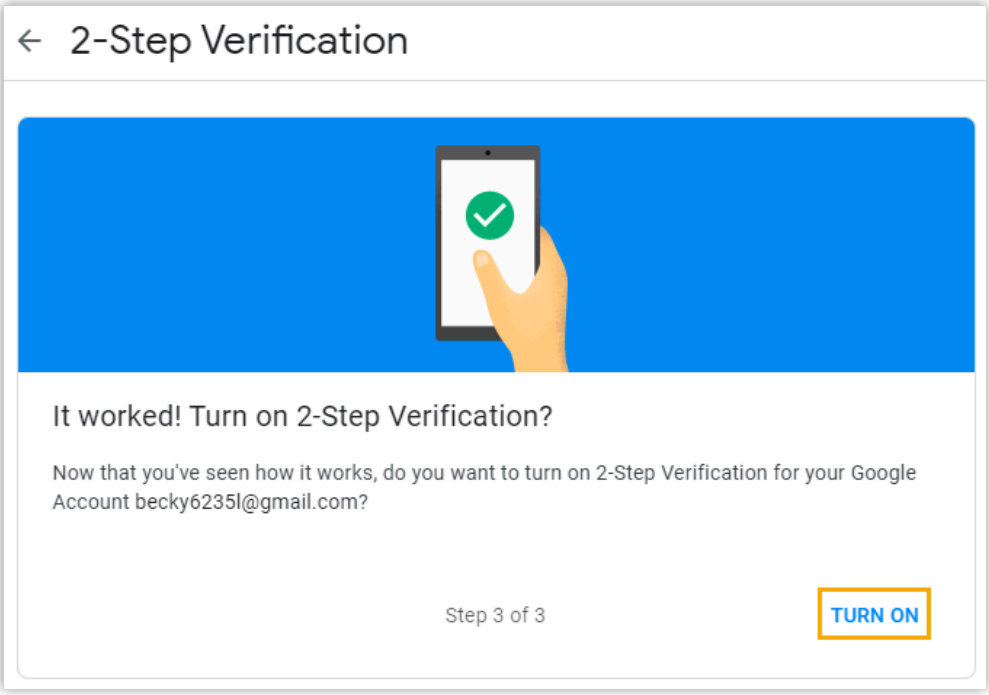
Prerequisites

Make sure Yeastar P-Series Cloud Edition can access Google Server.

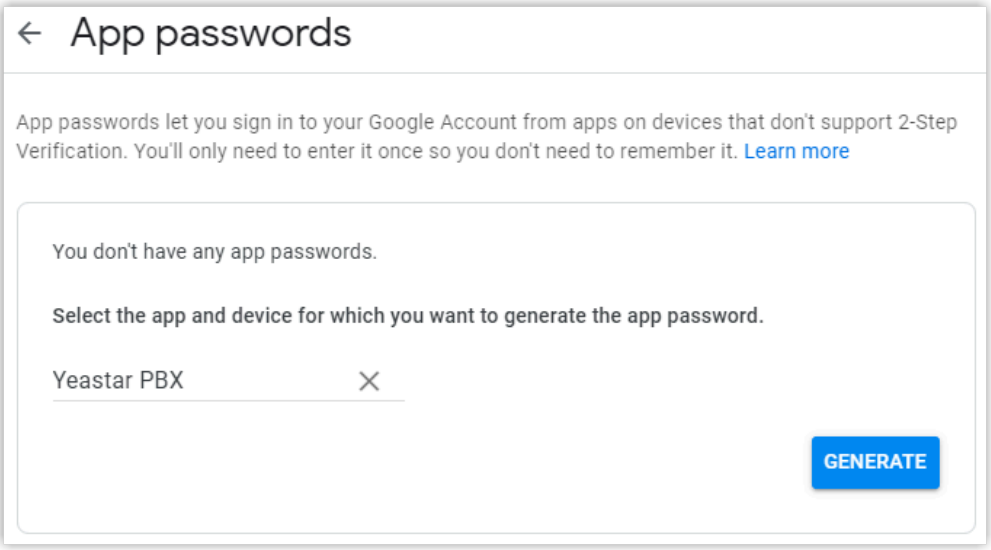
Step1. Create an app password on Google Account

To ensure that the PBX can access Gmail server, you need to turn on 2-Step verification and create an app password as follows.

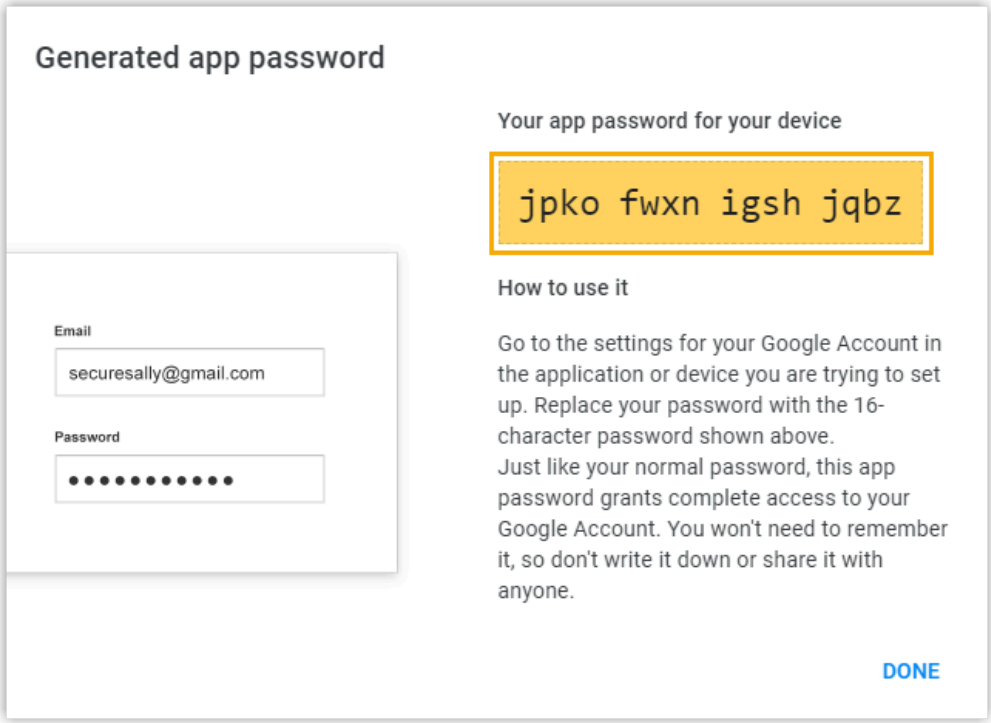
1. Sign in to Google Account by your Gmail account.
2. On the left navigation bar, click **Security**.
3. Turn on 2-Step Verification.
    a. In the **Signing in to Google** section, click **2-Step Verification** and enter your Gmail password to verify your account.
    b. On the **2-Step Verification** page, click **GET STARTED** and enter your Gmail password to verify your account.
    c. Select a verification method, verify your account according to the prompt.
    d. Click **TURN ON** to turn on 2-step verification.



4. Right above the page, click ← to back to the security page.
5. Create an app password.
    a. In the **Signing in to Google** section, click **App passwords** and enter your Gmail password to verify your account.
    b. In the **Select app** drop-down list, select **Other (Custom name)**.
    c. In the text field, enter a name to help you identify the app password. For example, enter *Yeastar PBX*.
    d. Click **GENERATE**.



An app password is generated. Note down the password, which is used to verify your Gmail account when you configure Gmail as the mail server in the PBX.



Step2. Configure Gmail as mail server of Yeastar P-Series Cloud Edition

To ensure that the PBX can access Gmail server via your Google account, you need to proceed as follows:

1. Log in to PBX web portal, go to **System > Email > Email Server**.
2. In the **Type of Email Server** drop-down list, select **Custom Email Server**.
3. In the **Select Email Server Provider** drop-down list, select **General**.
4. Configure email server settings.
    - **Sender Email Address**: Enter your Gmail address, which will appear as the From address for outgoing emails sent by the PBX.
    - **Email Address or Username**: Enter your Gmail address.
    - **Password**: Enter the 16-digit app password, which is used to access Gmail server.
    - **Outgoing Mail Server (SMTP)**: Retain the default value *smtp.gmail.com*.
    - **Port**: Retain the default value *587*.
    - **Enable TLS Encryption**: Keep the option unselected.
5. Test if the mail server can successfully send emails.
    a. Click **Test**.
    b. In the pop-up window, enter a recipient's email address in the **Email Address** field.
    c. Click **Test**.
6. Click **Save**.

Result

- If the test email is sent successfully, the page displays "Success" and the recipient's mailbox would receive the email.
- If the test email is failed to be sent, the page displays "Failed to send" and prompts you an error message. You can check the error in Email Sent Logs.

Set up Outlook as an Email Server

This topic describes how to set up Microsoft Outlook as the email server for Yeastar P-Series Cloud Edition using modern authentication. This allows the PBX to send and receive emails through your Microsoft account.

Requirements

Platform	Requirement
PBX server	• **Firmware**: Version 84.14.0.24 or later • **Plan**: Enterprise Plan (EP) or Ultimate Plan (UP)
Microsoft 365	• The Microsoft account should meet the following requirements:

Platform	Requirement
	Note: The account will be used as the designated email sender for the SMTP server. ◦ Assigned with the **Global Administrator** role. ◦ Assigned with **Microsoft 365 Business Standard** license. • The modern authentication is enabled in Microsoft 365 admin center.

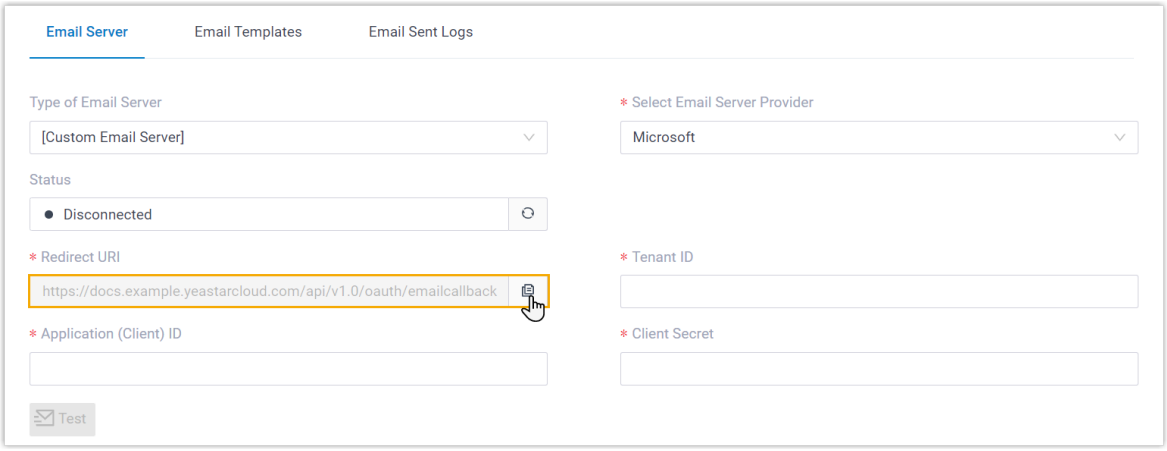
Procedure

- Step 1. Obtain redirect URI from PBX
- Step 2. Register an application on Microsoft Entra ID
- Step 3. Configure SMTP settings on PBX

Step 1. Obtain redirect URI from PBX

Obtain the redirect URI from Yeastar P-Series Cloud Edition, which will be required when registering a Microsoft Entra application later.

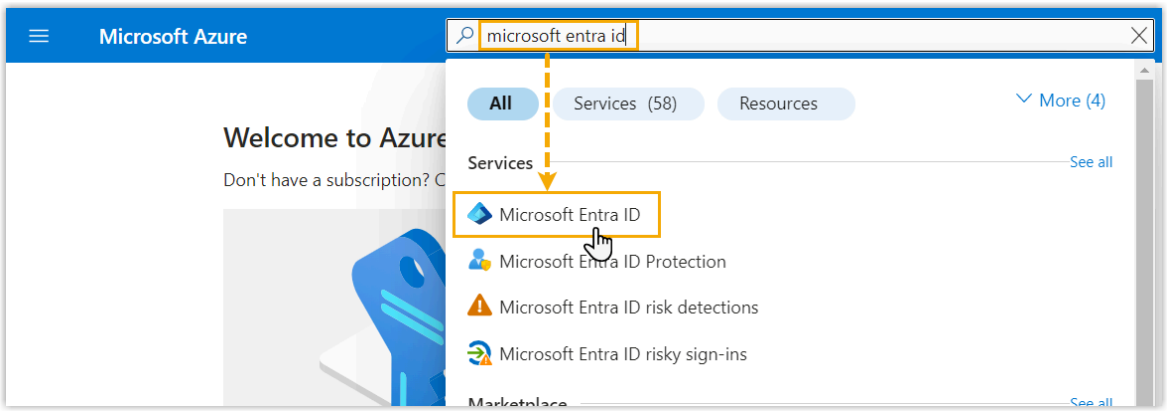
1. Log in to PBX web portal, go to **System > Email > Email Server**.
2. In the **Type of Email Server** drop-down list, select **Custom Email Server**.
3. In the **Select Email Server Provider** drop-down list, select **Microsoft**.
4. In the **Redirect URI** field, click to copy the URI.



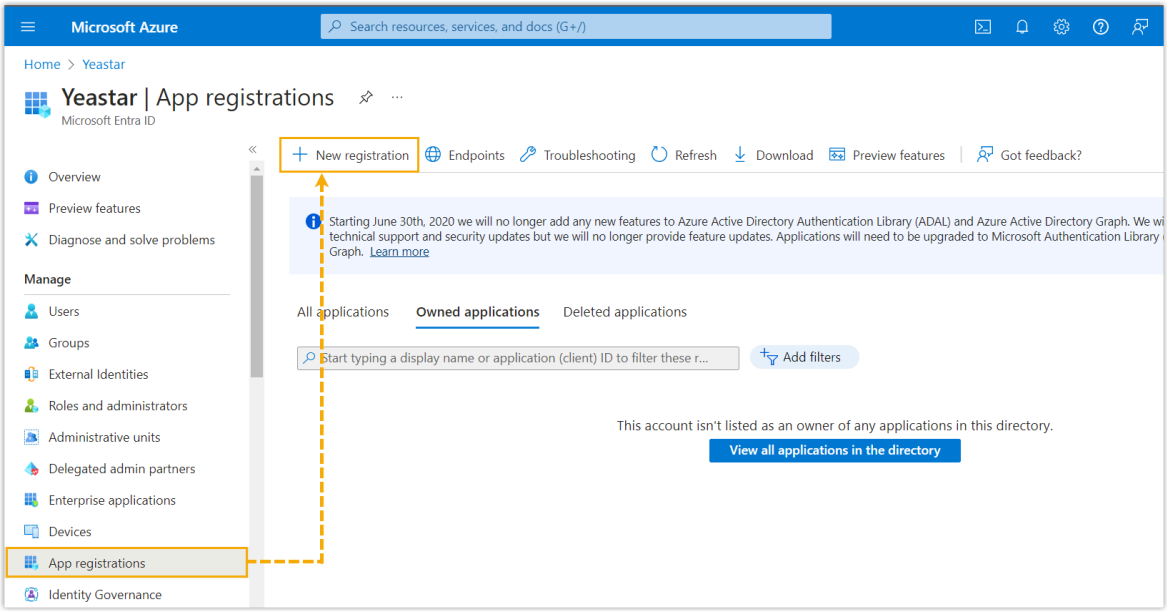
Step 2. Register an application on Microsoft Entra ID

Register an application on Microsoft Entra ID with the redirected URI provided by Yeastar PBX, then obtain the application authentication information for SMTP settings on PBX.

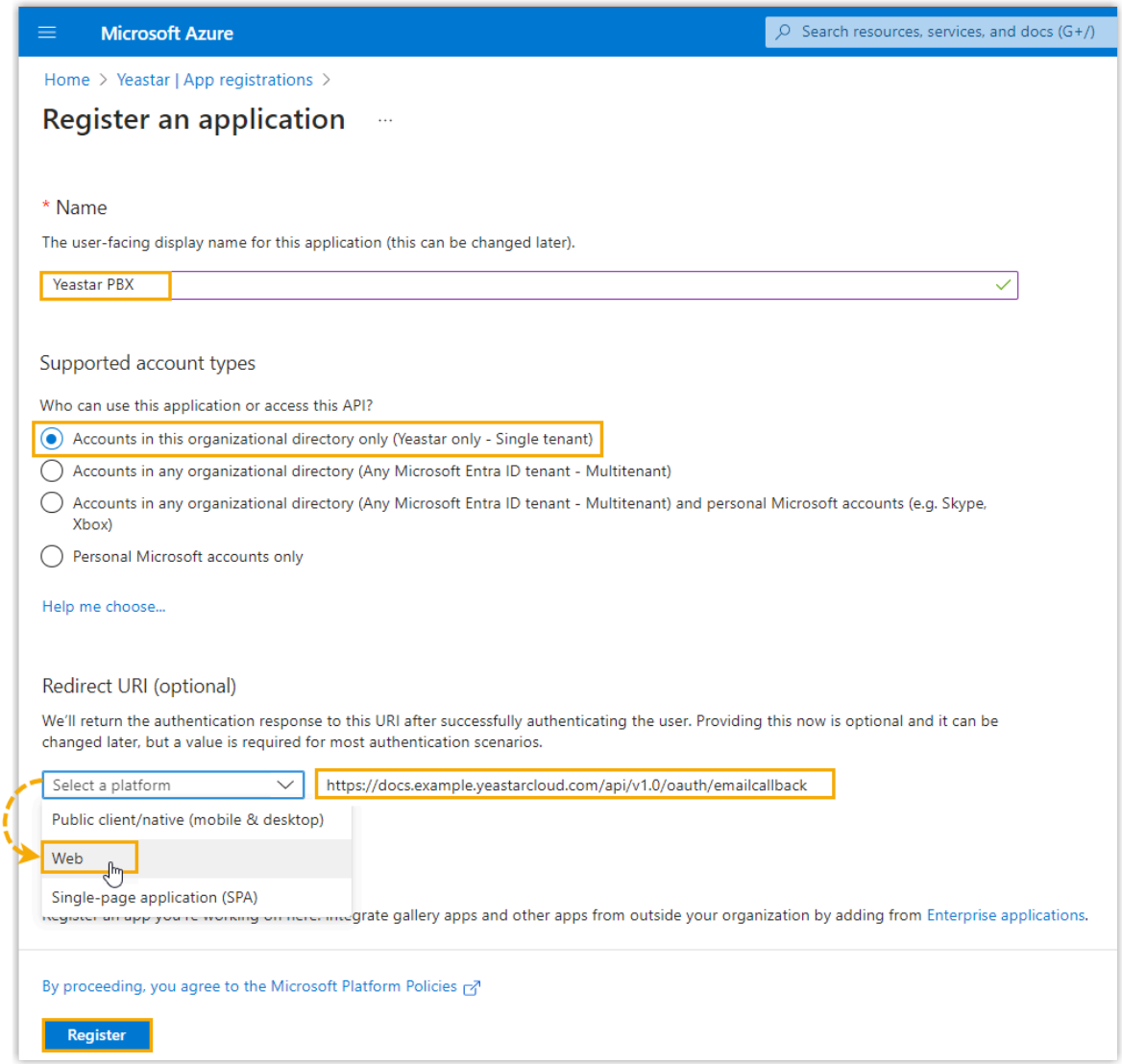
1. Log in to Microsoft Azure Portal with the Microsoft Administrator account.
2. In the search bar, search and select **Microsoft Entra ID** service to enter your organization's directory.



3. On the left navigation bar of organization's directory, go to **App registrations**, then click **New registration**.



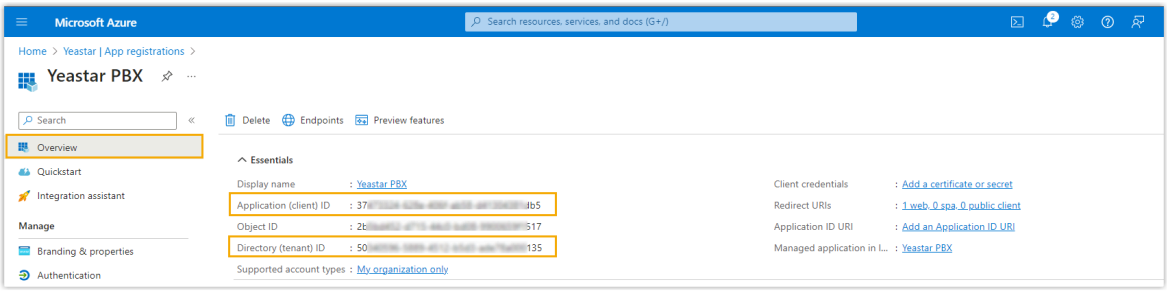
4. In the **Register an application** page, configure the registration information for the application, and click **Register**.



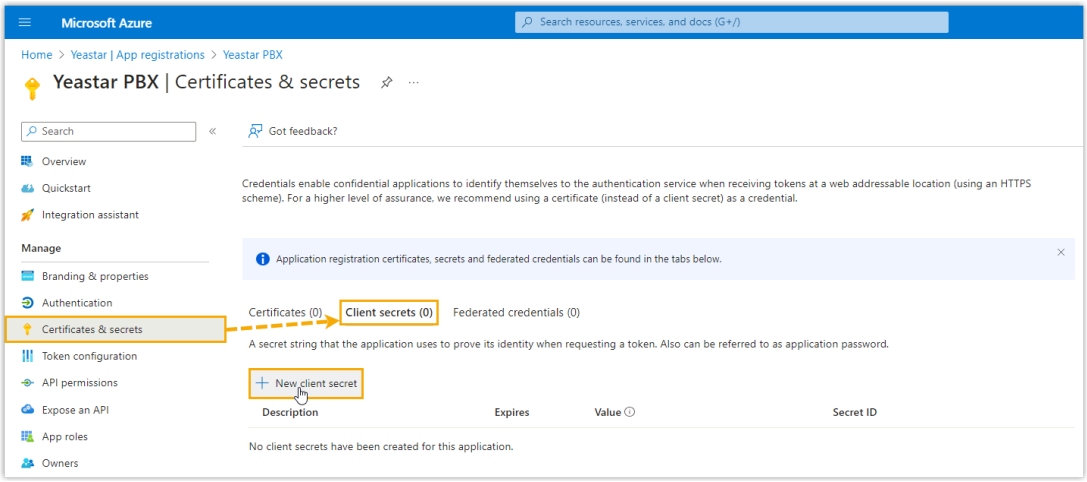
- **Name**: Specify a name to help you identify the application.
- **Supported account types**: Select **Accounts in this organizational directory only**.
- **Redirect URI**: In the **Select a platform** drop-down list, select **Web**, then paste the Redirect URI obtained from the PBX.

An application is registered successfully, you are redirected to the **Overview** page of the application.

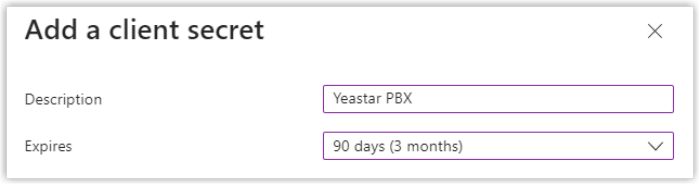
5. Note down the **Application (client) ID** and **Directory (tenant) ID** of the application, as you will need to fill them into the PBX later.



6. Generate a client secret for the application.
    a. On the left navigation bar of the application, go to **Certificates & secrets > Client secrets**, then click **New client secret**.

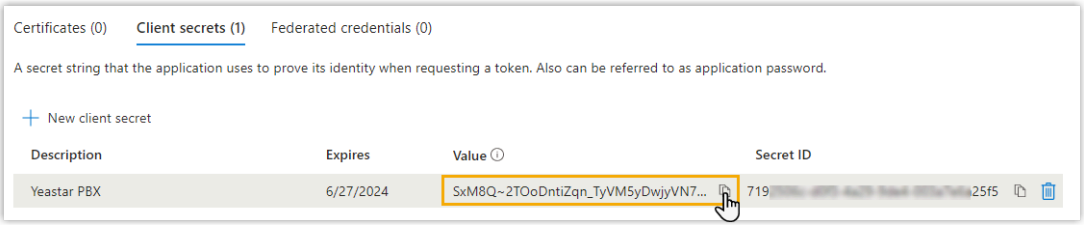


    b. In the **Add a client secret** page, add a description and set an expiration date for the client secret, then click **Add**.



    A client secret is created and displayed in the **Client secrets** list.
    c. Note down the client secret's **Value** as you will need to fill it into the PBX later.

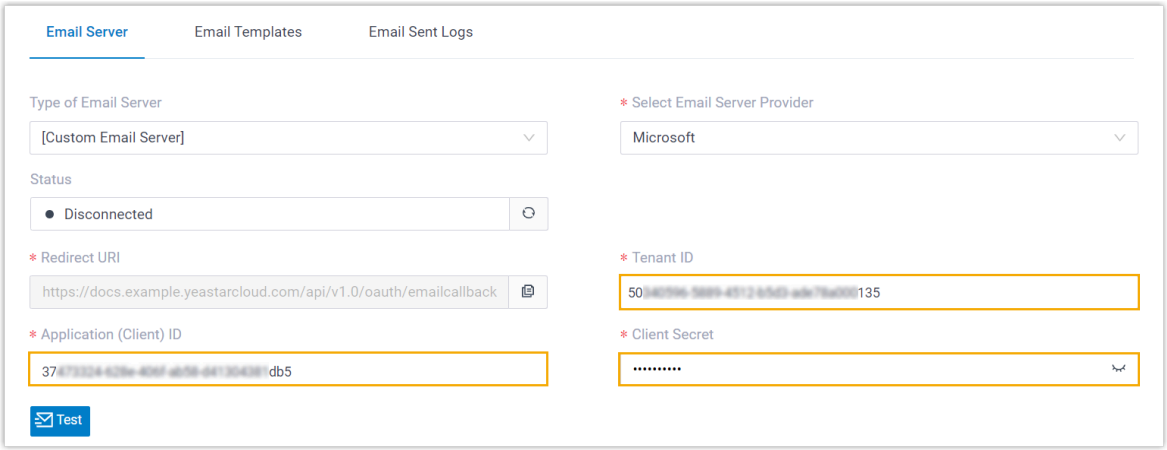
> **Important:**
> Record the client secret's value before leaving the page, as the key is only shown once. Otherwise, you will have to create a new client secret.



Step 3. Configure SMTP settings on PBX

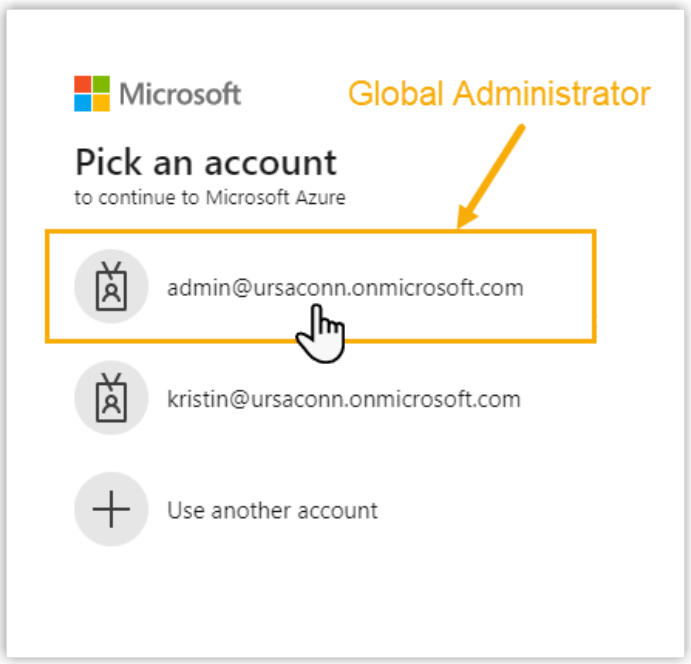
Configure the SMTP settings of the PBX with the authentication information obtained from the Microsoft Entra application, so as to set Microsoft Outlook as the PBX's email server.

1. Log in to PBX web portal, go to **System > Email > Email Server**.
2. In the **Type of Email Server** drop-down list, select **Custom Email Server**.
3. In the **Select Email Server Provider** drop-down list, select **Microsoft**.
4. Enter the following authentication information.

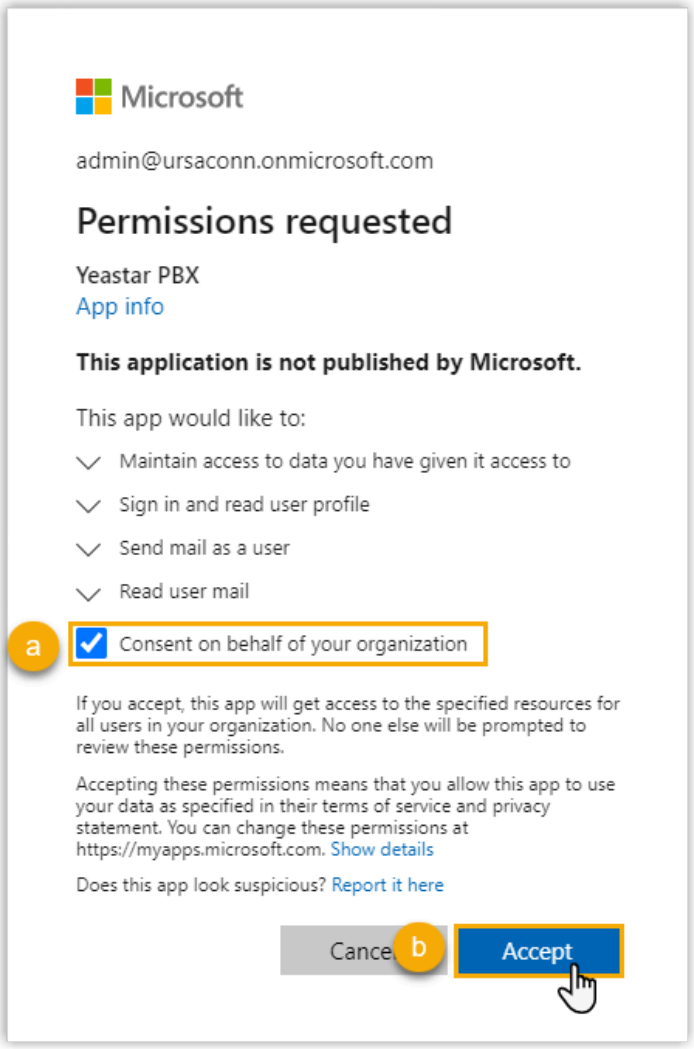


   - **Tenant ID**: Paste the Tenant ID of the Microsoft Entra application.
   - **Application (Client) ID**: Paste the Application ID of the Microsoft Entra application.
   - **Client Secret**: Paste the client secret of the Microsoft Entra application.
5. Click **Save**.

   You are redirect to the Microsoft Sign-in page.
6. Sign in with the Microsoft account that has **Global Administrator** privilege.



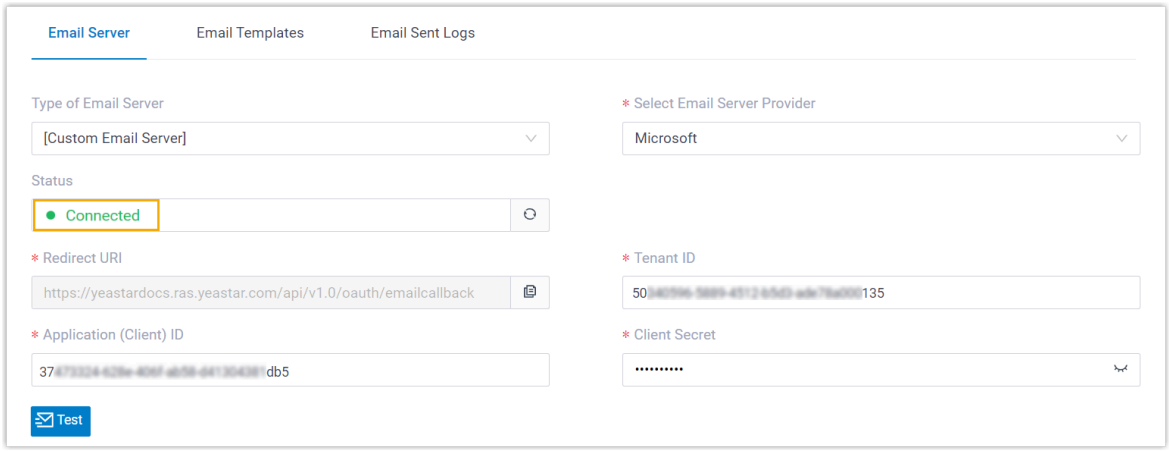
7. In the pop-up window, do as follows to confirm:



a. Select the checkbox of **Consent on behalf of your organization**.

b. Click **Accept**.

On the PBX configuration page, the **Status** field displays **Connected**, indicating that the Microsoft authentication is successful; The Microsoft Outlook is set up as the email server.



8. Test if the mail server can successfully send emails.
    a. Click **Test**.
    b. In the pop-up window, enter a recipient's email address in the **Email Address** field, then click **Test**.

Result

A test email is sent successfully using the Microsoft account; The page displays "Success" and the recipient's mailbox would receive the email where the sender name and email address are the Microsoft account's display name and user principal name respectively. For example, `admin<admin@ursaconn.onmicrosoft.com>`.

Note:

- If you want to change the sender name of the emails sent by the SMTP server, change the display name of the Microsoft account.
- If the test email is failed to be sent, the page displays "Failed to send" and prompts you an error message.

Customize Email Templates

This topic describes how to customize email notification language and email templates.

Background information

If you have enabled notification for a specific event, and have chosen to send emails to notify contacts, the system will send emails in the pre-configured email template to inform contacts when the event is triggered.

Yeastar P-Series Cloud Edition provides the following types of email templates:

- **Operations**: Changes of password and login status.
- **Telephony**: SIP trunk registration and emergency calling.
- **System**: System performance, such as new system firmware detected, system upgrade completed, etc.
- **Security**: Such as web login block, auto defense, etc.
- **Event Reminder**: Reminders related with the subscribed plan and services.
- **Email**: Email notifications related with extensions.

Procedure

1. Log in to PBX web portal, go to **System > Email > Email Templates**.
2. Set the language of notification emails.

   > **Note:**
   > If you fail to find the desired language, you can update templates based on **English**.

   a. Click **Notification Email Language**.
   b. In the pop-up window, select a language from the drop-down list.
   c. Click **Save**.

   The system will send emails in the selected language.
3. Edit a desired email template.

   a. In the **Email Templates** list, click  beside the desired email template.
   b. In the **Template** drop-down list, select **Custom**.
   c. Edit email subject and content according to your needs.

   > **Note:**
   > Images, videos, and audios are not supported.
4. Click **Save** and **Apply**.

Email Sent Logs

This topic introduces email sent logs and describes how to query logs.

Email sent logs overview

Email sent logs allow you to monitor mail delivery and provide you with error messages to help you troubleshoot delivery issues more quickly.

Auto cleanup of email sent logs

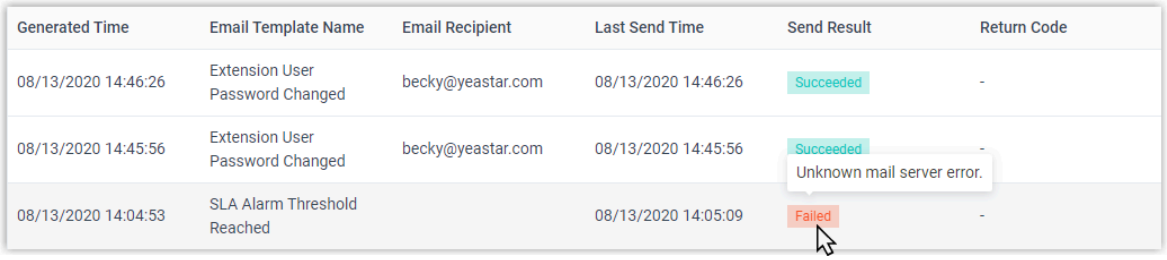
By default, when logs reach 50,000, the newest logs will replace the oldest logs. You can change the value, or restrict how long logs can be saved.

For more information, see Auto Cleanup Settings.

Query email sent logs

1. Log in to PBX web portal, go to **System > Email > Email Sent Logs**.
2. Query logs by the following criteria according to your needs.
   - **Send Result**: Query all logs or query logs by send result.
   - **Email Template Name**: Query logs by email template.
   - **Generated Time**: Query logs by the generated date and time.
   - **Email Recipient**: Query logs by emails' recipients.

   After logs are filtered, you can hover your mouse over **Failed** beside the failed log to check the error message.



Storage

Auto Cleanup Settings

Data of Yeastar P-Series Cloud Edition is stored in a default storage area. Yeastar P-Series Cloud Edition provides an Auto Cleanup feature, which can automatically and periodically cleans up your CDR, voicemails, backup files, and logs (including event logs, email sent logs, operation logs, and system logs). This topic describes relevant configuration parameters of auto cleanup.

CDR Auto Cleanup

Table 36.

Setting	Description
Max Number of CDR	Set the maximum number of CDR that should be retained. When it reaches 90% of the maximum number, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum number, the oldest CDR will be deleted. **Default value**: 200,000 **Maximum value**: 1,000,000
CDR Preservation Days	Set the maximum number of days that CDR should be retained. When it reaches 90% of the maximum preservation days, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum preservation days, the oldest CDR will be deleted. **Default value**: 0, which means no limit.

Voicemail Auto Cleanup

Table 37.

Setting	Description
Max Number of Voicemail	Set the maximum number of voicemails that should be retained. When it reaches 90% of the maximum number, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum number, the oldest voicemails will be deleted. **Default value**: 100 **Maximum value**: 500
Voicemail Preservation Days	Set the maximum number of days that voicemails should be retained. When it reaches 90% of the maximum preservation days, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods.

Table 37. (continued)

Setting	Description
	When it reaches the maximum preservation days, the oldest voicemails will be deleted. **Default value**: 0, which means no limit.

External Chat Data Auto Cleanup

Setting	Description
Max Number of External Chat Data	Set the maximum number of external chat data to store. When it reaches 90% of the maximum number, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum number, the oldest external chat data will be deleted. **Default value**: 50,000 **Maximum value**: 100,000
External Chat Data Preservation Days	Set the maximum number of days that external chat data should be retained. When it reaches 90% of the maximum preservation days, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum preservation days, the oldest external chat data will be deleted. **Default value**: 0, which means no limit.

System Backup Files

Table 38.

Setting	Description
Max Number of Files	Set the maximum number of backup files that should be retained. When it reaches 90% of the maximum number, the **Auto Cleanup Reminder** event will be triggered, the system will notify contacts concerned via specific methods. When it reaches the maximum number, the oldest backup files will be deleted. **Default value**: 5

Table 38. (continued)

Setting	Description
	Maximum value: 8

Event Logs Auto Cleanup

Table 39.

Setting	Description
Max Number of Logs	Set the maximum number of event logs that should be retained. When it reaches the maximum number, the oldest event logs will be deleted. **Note:** Some event logs are not allowed to be automatically cleaned up. When those logs reach 5,000, the oldest event logs will be deleted. **Default value**: 50,000 **Maximum value**: 1,000,000
Logs Preservation Days	Set the maximum number of days that event logs should be retained. When it reaches the maximum preservation days, the oldest event logs will be deleted. **Default value**: 0, which means no limit.

Email Sent Logs Auto Cleanup

Table 40.

Setting	Description
Max Number of Logs	Set the maximum number of email sent logs that should be retained. When it reaches the maximum number, the oldest email sent logs will be deleted. **Default value**: 50,000 **Maximum value**: 1,000,000
Logs Preservation Days	Set the maximum number of days that email sent logs should be retained. When it reaches the maximum preservation days, the oldest email sent logs will be deleted.

Table 40. (continued)

Setting	Description
	Default value: 0, which means no limit.

Operation Logs Auto Cleanup

Table 41.

Setting	Description
Max Number of Logs	Set the maximum number of operation logs that should be retained. When it reaches the maximum number, the oldest operation logs will be deleted. **Default value**: 50,000 **Maximum value**: 1,000,000
Logs Preservation Days	Set the maximum number of days that operation logs should be retained. When it reaches the maximum preservation days, the oldest operation logs will be deleted. **Default value**: 0, which means no limit.

System Logs Auto Cleanup

Table 42.

Setting	Description
Max Storage of Logs (MB)	Set the maximum file size for a system log package. When it reaches the maximum file size, the oldest logs in the package will be deleted. **Default value**: 10
Logs Preservation Days	Set the maximum number of days that a system log should be retained. **Default value**: 7 **Maximum value**: 15

Archive

Yeastar P-Series Cloud Edition Remote Archiving Overview

Yeastar P-Series Cloud Edition allows you to download and archive the system's call recordings and backup files to external servers via FTP, SFTP, Amazon S3, and Google Cloud Storage services, enabling less space occupying and minimized data loss risk on PBX, as well as easier file management on external server.

Requirements and restrictions

Requirements

- **PBX Firmware**: Version 84.15.0.22 or later
- **PBX Plan**: Enterprise Plan (for self-managed phone system) or Ultimate Plan (for Yeastar hosted phone system)

Restrictions

- **Archive server**: 10
- **Archive task**: 200

Archive Files to FTP Server

Yeastar P-Series Cloud Edition supports archiving the system's call recordings and backup files to FTP server, either on a regular interval or at any time you want. This topic describes how to add FTP server as an archive server and schedule tasks to archive the desired files.

Requirements and restrictions

Requirements

- **PBX Firmware**: Version 84.15.0.22 or later
- **PBX Plan**: Enterprise Plan (for self-managed phone system) or Ultimate Plan (for Yeastar hosted phone system)

Restrictions

- **Archive server**: 10
- **Archive task**: 200

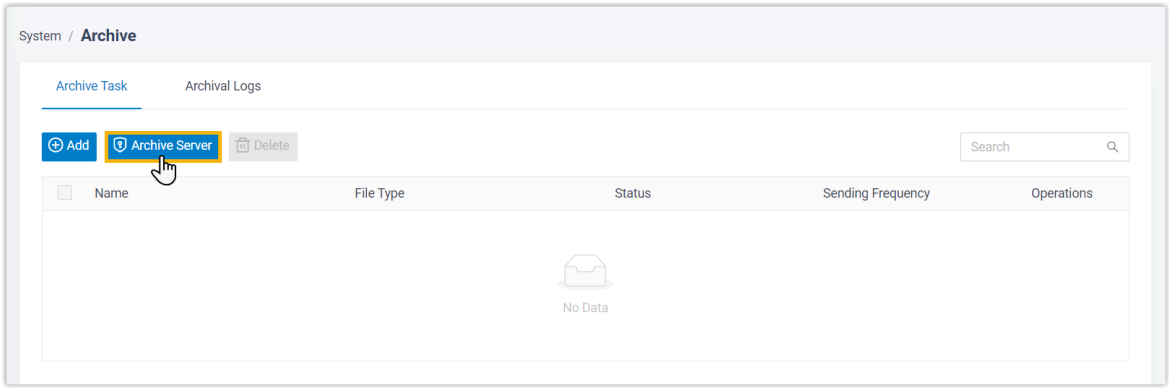
Prerequisites

Before you begin, you need to prepare the following resources and collect the required information:

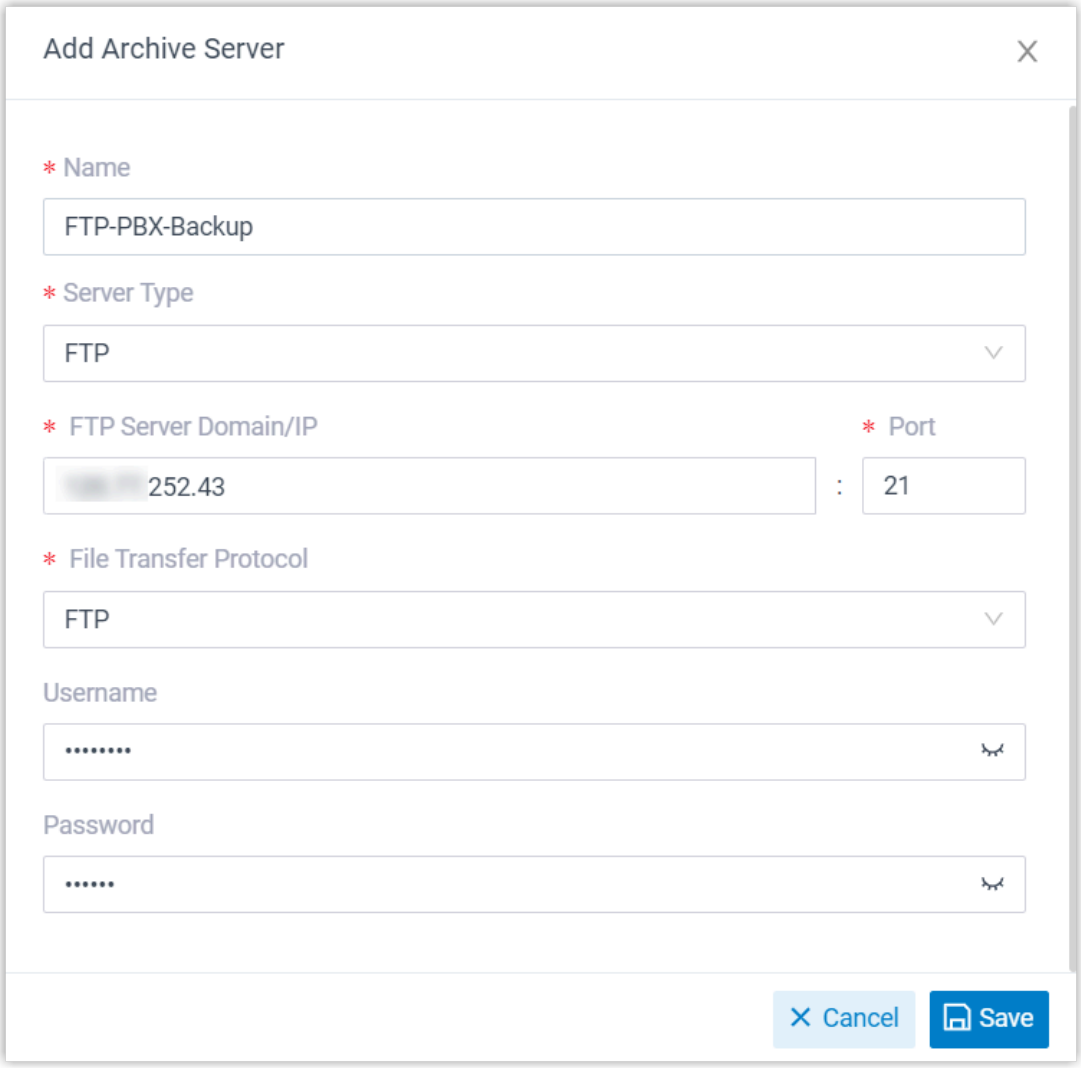
- Prepare an FTP server, and note down its **Domain/IP Address**, **Port**, and **File Transfer Protocol**.
- Prepare an FTP account with both read and write permissions, and note down its **Username** and **Password**.

Step 1. Add FTP server as archive server

1. Log in to PBX web portal, go to **System > Archive**.
2. Under the **Archive Task** tab, click **Archive Server**.



3. Set up FTP server as the archive server.
    a. Click **Add**.
    b. In the pop-up window, complete the following settings.



Setting	Description
Name	Enter a name to help you identify the server.
Server Type	Select **FTP**.
FTP Server Domain/IP	Enter the address of your FTP server.
Port	Enter the port on which your FTP server is running. The default port is 21.
File Transfer Protocol	Select the transfer protocol supported by the FTP server. • **FTP**: Standard File Transfer Protocol. • **FTPES**: FTP over Explicit TLS/SSL. **Note:**

Setting	Description
	◦ The TLS feature on the FTP server must be enabled, and the encryption policy must be well-configured for security reason. ◦ The supported TLS protocol version is **TLS V1.2**.
Username	Enter the username for the FTP account.
Password	Enter the password associated with the username.

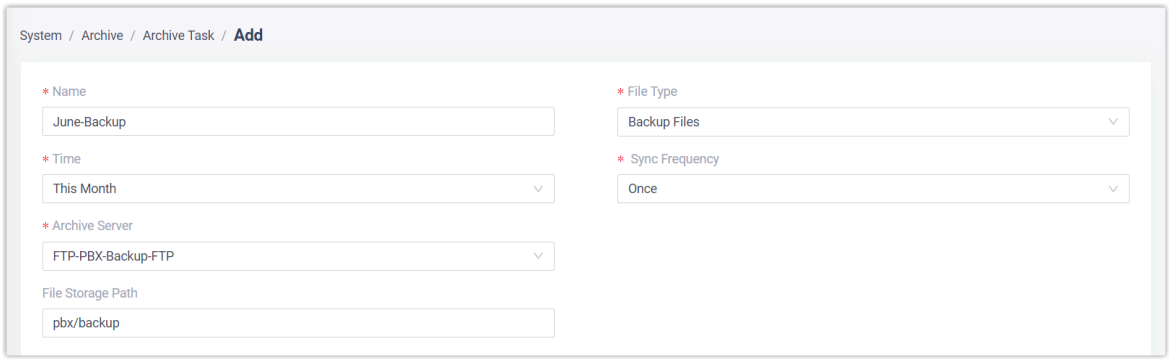
c. Click **Save**.

The FTP server is added as an archive server and displayed on the archive server list.

4. Click ✕ to close the window.

Step 2. Create a task to archive files to FTP server

1. Under **Archive Task** tab, click **Add**.
2. Create a one-time or recurring archive task.



Setting	Description
Name	Enter a name to help you identify the task.
File Type	Select **Recording Files** or **Backup Files** as needed.
Time	Specify a time range of the files to be archived. **Note:** You can archive files for up to 31 days at a time.
Sync Frequency	Set how often to archive files to FTP server. **Note:**

Setting	Description
	As large amounts of data will consume PBX's CPU resources, we recommend that you schedule archive tasks during off-peak hours. • **Once**: If you choose the option, the system will archive files immediately after you save the task. • **Daily**: If you choose the option, select a time from the drop-down list. The system will archive files at this time of the day. • **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system will archive files at this time of the week. • **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system will archive files on this day and time of the month.
Archive Server	Select the FTP server that you have added.
Select Folder/Path	Optional. Set the path to the folder in which you want to store archived files. For example, `pbx/backup/`. **Note:** If you leave this field blank, the files will be stored under the root directory.

3. Click **Save**.

Result

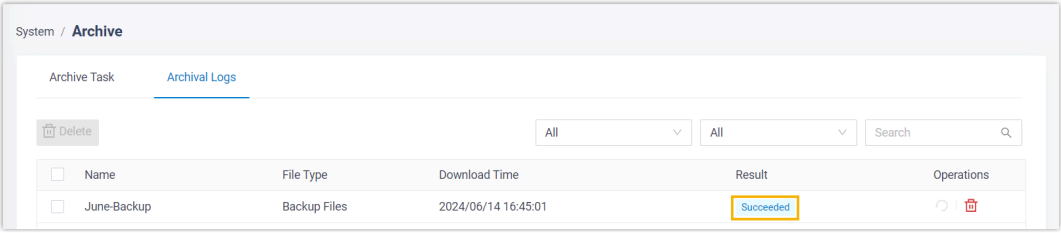
The specified files will be archived to the designated folder in your FTP server immediately or at the scheduled time.

Note:
The system executes only one task at a time to avoid affecting system performance. If there are multiple tasks, they will be queued up one after another.

You can check the archive result in the following ways.

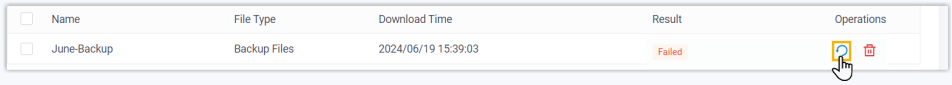
Check the archive result on PBX

On PBX web portal, go to **System > Archive > Archival Logs**. If the **Result** column of the task shows **Succeeded**, it indicates that the specified files have been successfully archived to the FTP server.



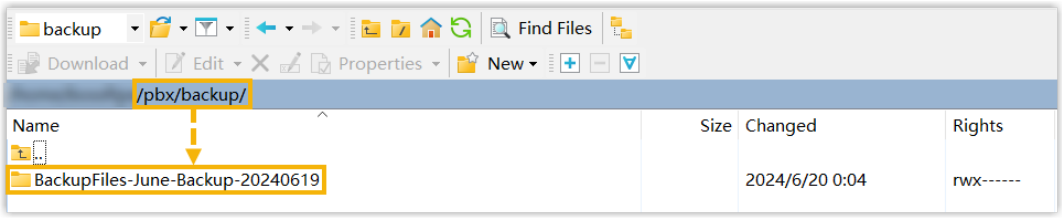
Note:

If the task is failed, the **Failed to Archive File(s)** event will be triggered; You can click to retry the task.



Check the archive result on FTP server

Go to the designated folder in FTP server. If the specified files appear in the list, it indicates that the archive is successful.



Archive Files to SFTP Server

Yeastar P-Series Cloud Edition supports archiving the system's call recordings and backup files to SFTP server, either on a regular interval or at any time you want. This topic describes how to add SFTP server as an archive server and schedule tasks to archive the desired files.

Requirements and restrictions

Requirements

- **PBX Firmware**: Version 84.15.0.22 or later
- **PBX Plan**: Enterprise Plan (for self-managed phone system) or Ultimate Plan (for Yeastar hosted phone system)

Restrictions

- **Archive server**: 10
- **Archive task**: 200

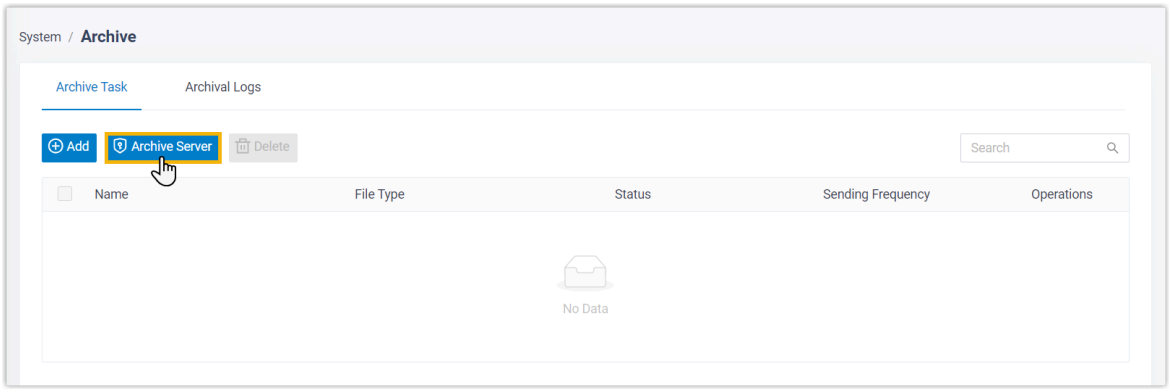
Prerequisites

Before you begin, you need to prepare the following resources and collect the required information:

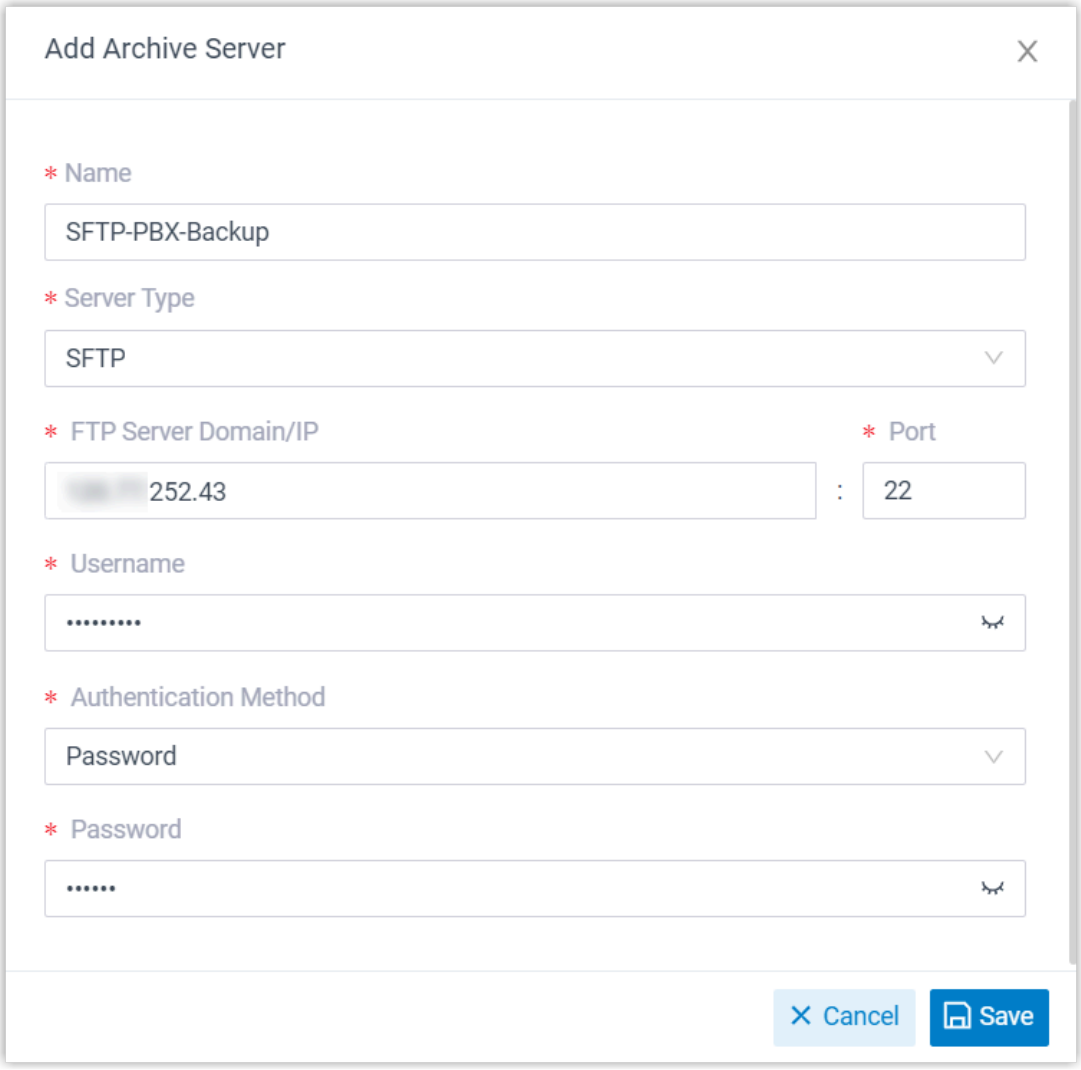
- Prepare an SFTP server and note down its **Domain/IP Address** and **Port**.
- Prepare an SFTP account with both read and write permissions, and note down its **Username** and **Password/Private Key**.

Step 1. Add SFTP server as archive server

1. Log in to PBX web portal, go to **System > Archive**.
2. Under the **Archive Task** tab, click **Archive Server**.



3. Set up SFTP server as the archive server.
   a. Click **Add**.
   b. In the pop-up window, complete the following settings.



Setting	Description
Name	Enter a name to help you identify the server.
Server Type	Select **SFTP**.
FTP Server Domain/IP	Enter the address of your SFTP server.
Port	Enter the port on which your SFTP server is running. The default port is 22.
Username	Enter the username for the SFTP account.
Authentication Method	Select an authentication method, then enter the credential.

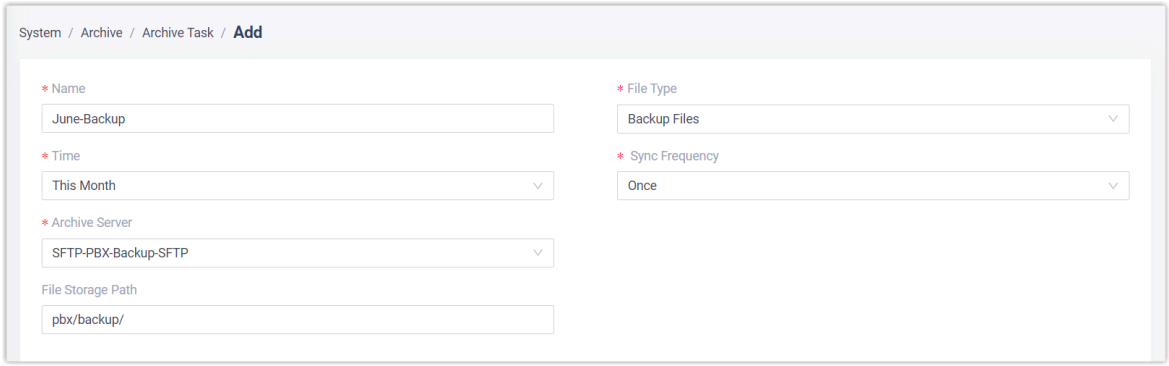
c. Click **Save**.

SFTP server is added as an archive server and displayed on the archive server list.

4. Click ✕ to close the window.

Step 2. Create a task to archive files to SFTP server

1. Under **Archive Task** tab, click **Add**.
2. Create a one-time or recurring archive task.



Setting	Description
Name	Enter a name to help you identify the task.
File Type	Select **Recording Files** or **Backup Files** as needed.
Time	Specify a time range of the files to be archived. **Note:** You can archive files for up to 31 days at a time.
Sync Frequency	Set how often to archive files to SFTP server. **Note:** As large amounts of data will consume PBX's CPU resources, we recommend that you schedule archive tasks during off-peak hours. • **Once**: If you choose the option, the system will archive files immediately after you save the task. • **Daily**: If you choose the option, select a time from the drop-down list. The system will archive files at this time of the day. • **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system will archive files at this time of the week.

Setting	Description
	• **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system will archive files on this day and time of the month.
Archive Server	Select the SFTP server that you have added.
Select Folder/Path	Optional. Set the path to the folder in which you want to store archived files. For example, `pbx/backup/`. **Note:** If you leave this field blank, the files will be stored under the root directory.

3. Click **Save**.

Result

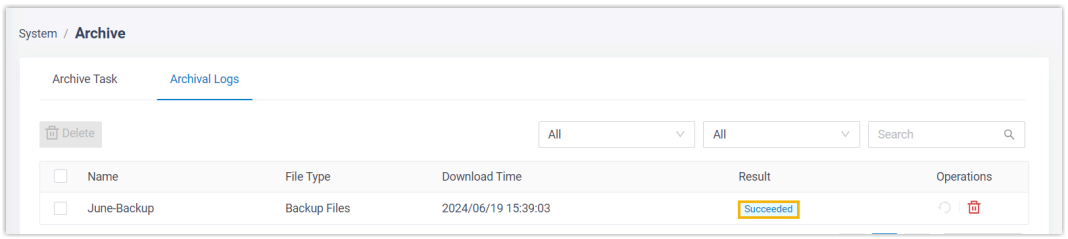
The specified files will be archived to the designated folder in your SFTP server immediately or at the scheduled time.

Note:
The system executes only one task at a time to avoid affecting system performance. If there are multiple tasks, they will be queued up one after another.

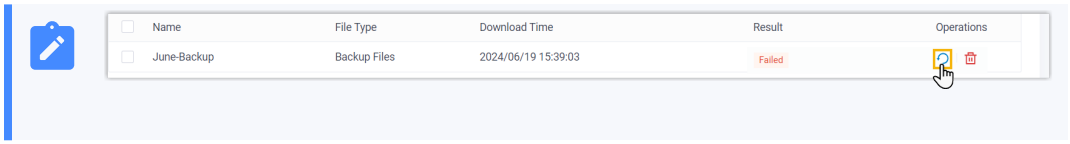
You can check the archive result in the following ways.

Check the archive result on PBX

On PBX web portal, go to **System > Archive > Archival Logs**. If the **Result** column of the task shows **Succeeded**, it indicates that the specified files have been successfully archived to SFTP server.

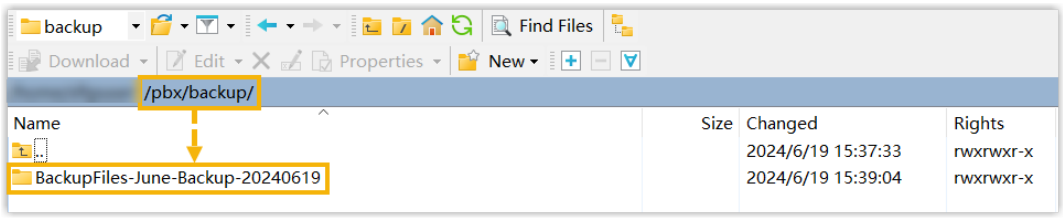


Note:
If the task is failed, the **Failed to Archive File(s)** event will be triggered; You can click to retry the task.



Check the archive result on SFTP server

Go to the designated folder in SFTP server. If the specified files appear in the list, it indicates that the archive is successful.



Archive Files to Amazon S3

Yeastar P-Series Cloud Edition supports archiving the system's call recordings and backup files to Amazon S3, either on a regular interval or at any time you want. This topic describes how to add Amazon S3 bucket as an archive server and schedule tasks to archive the desired files.

Requirements and restrictions

Requirements

- **PBX Firmware**: Version 84.15.0.22 or later
- **PBX Plan**: Enterprise Plan (for self-managed phone system) or Ultimate Plan (for Yeastar hosted phone system)

Restrictions

- **Archive server**: 10
- **Archive task**: 200

Introduction

To archive PBX's call recordings and backup files to a specific bucket in Amazon S3, you need to complete the followings:

1. Prepare an eligible account and bucket on AWS.

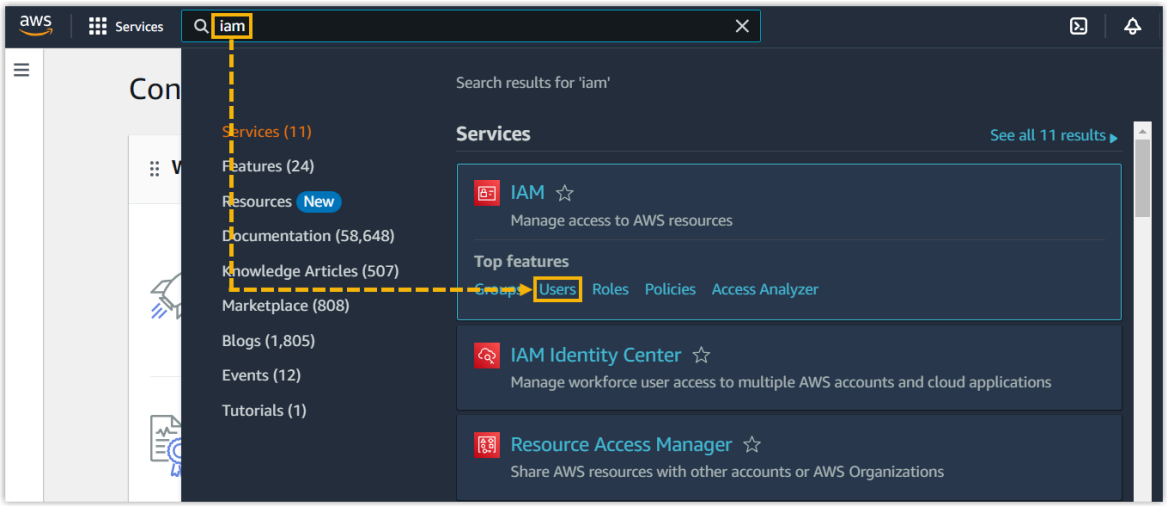
- An **IAM User** account that meets the requirements listed below:
  - Full access to Amazon S3 resources to archive PBX files
  - An access key pair (including an access key ID and a secret access key) to authenticate file archiving requests from PBX
- An **S3 bucket** to store archived PBX files

2. Set up archive server and task on PBX.
   - Add the Amazon S3 bucket as an archive server
   - Create tasks to archive PBX files to Amazon S3

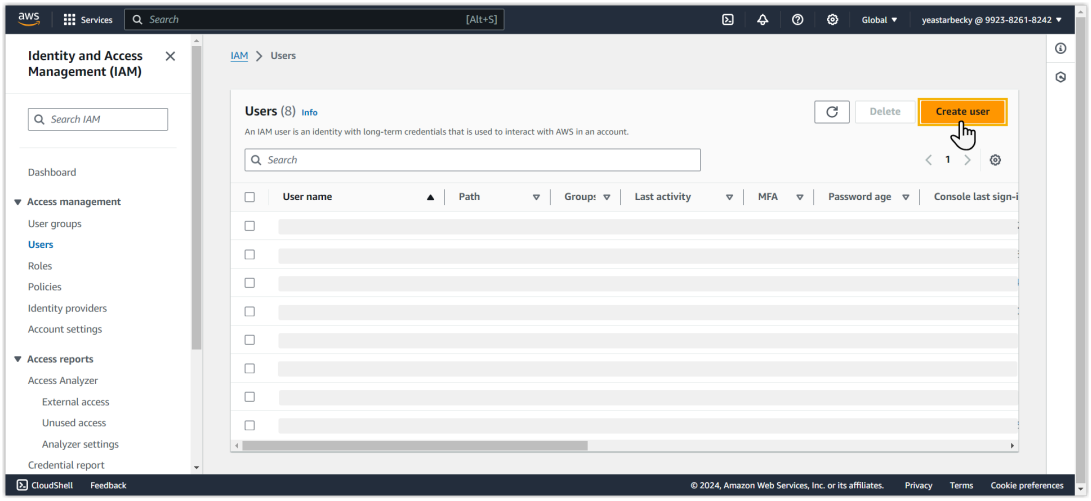
When it is time to execute the archive task, PBX will use the access key pair of the IAM user to send file archiving requests to Amazon S3. If the requests are considered authentic, the specified files will be archived to the designated bucket in Amazon S3.

Step 1. Create an IAM user with full access to Amazon S3 bucket on AWS

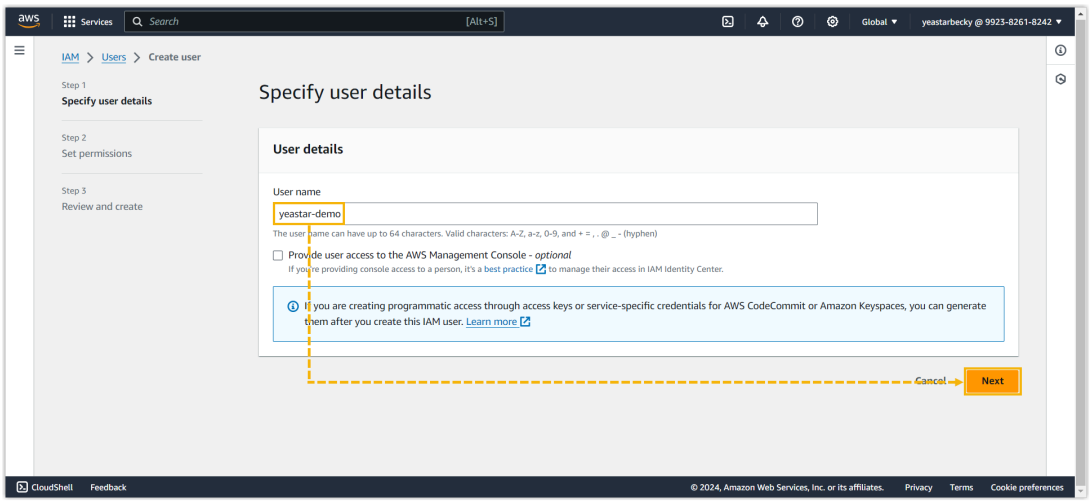
1. Log in to AWS Management Console.
2. At the top of the console, search for IAM service and select **Users**.



   You are redirected to the IAM users page.
3. Create an IAM user and grant the user full access to Amazon S3 bucket.
   a. At the top-right corner, click **Create user**.

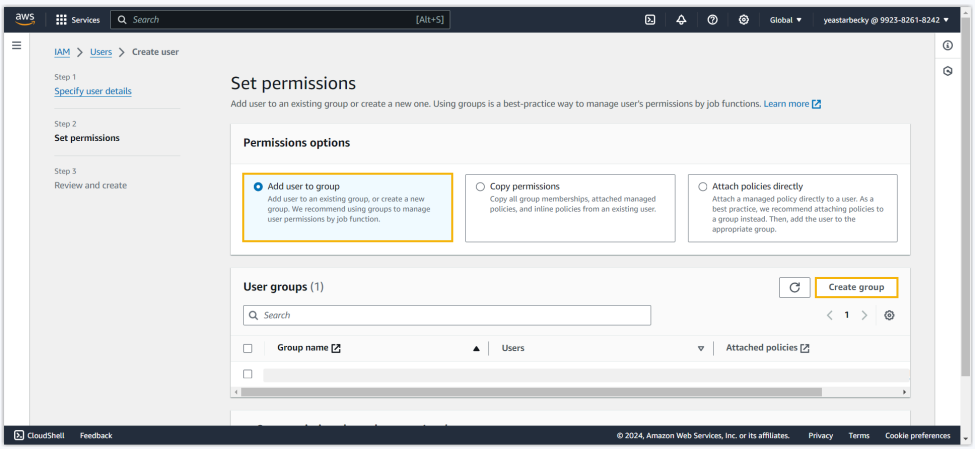


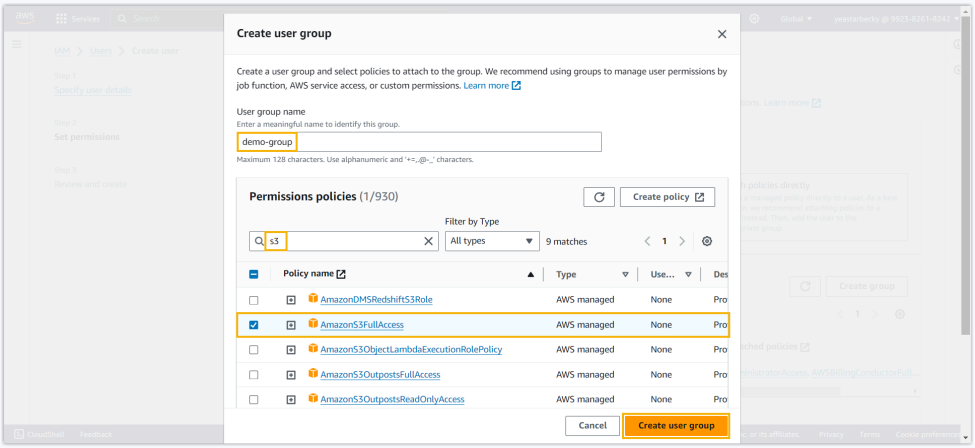
b. On the **Specify user details** page, enter a name in the **User name** field to help you identify the IAM user, then click **Next**.

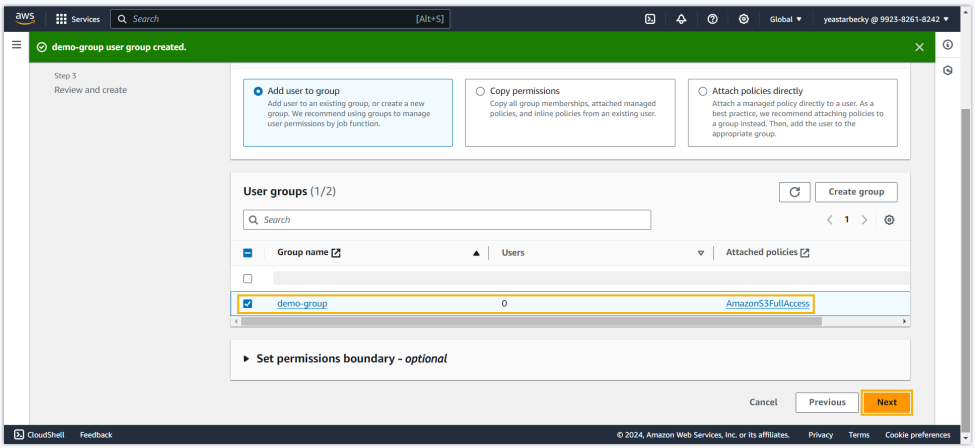


c. On the **Set permissions** page, grant the user full access to Amazon S3 buckets, then click **Next**.

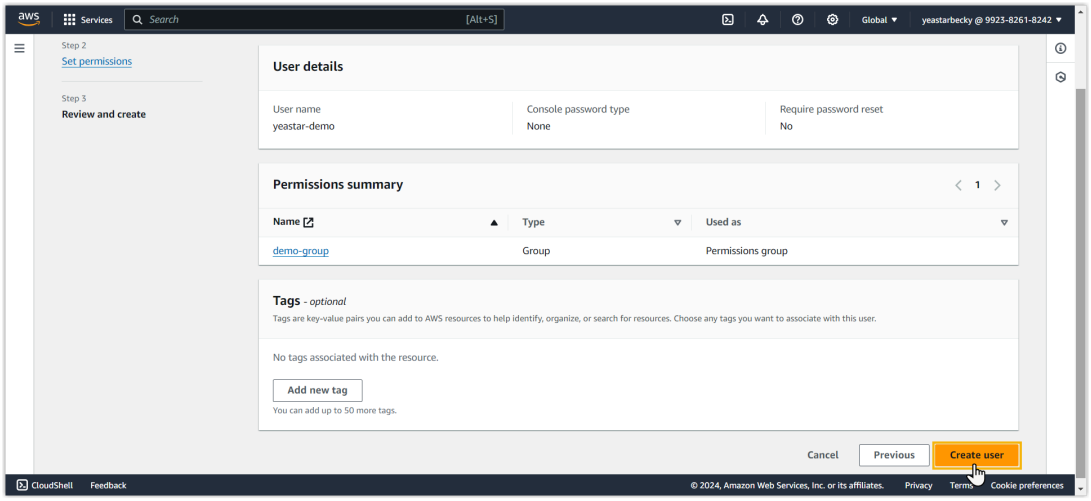
> **Note:**
> You can grant permission to the user by adding the user to an existing group, creating a new group, or attaching permission policy directly to the user.
>
> In this example, we create a new group with full Amazon S3 access permission and add the user to the group, as the following figure shows.





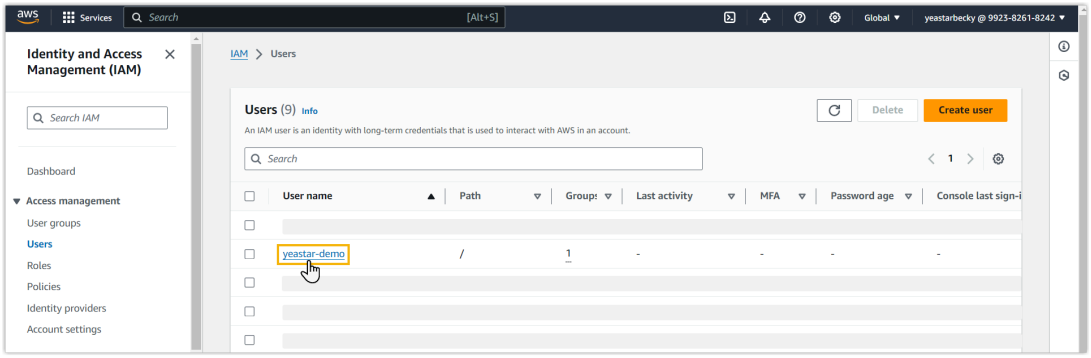


d. On the **Review and create** page, click **Create user**.

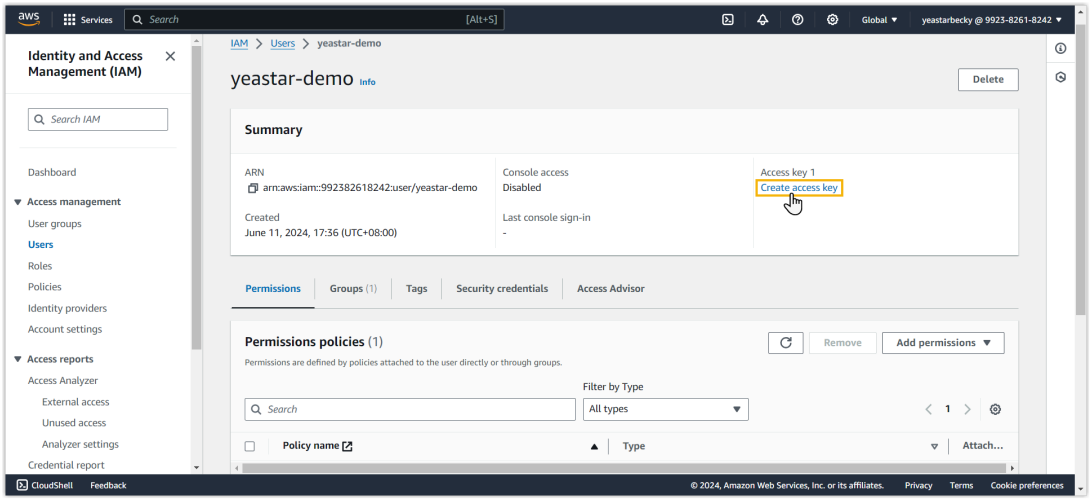


The IAM user with full access to Amazon S3 bucket is created and displayed on the users list.

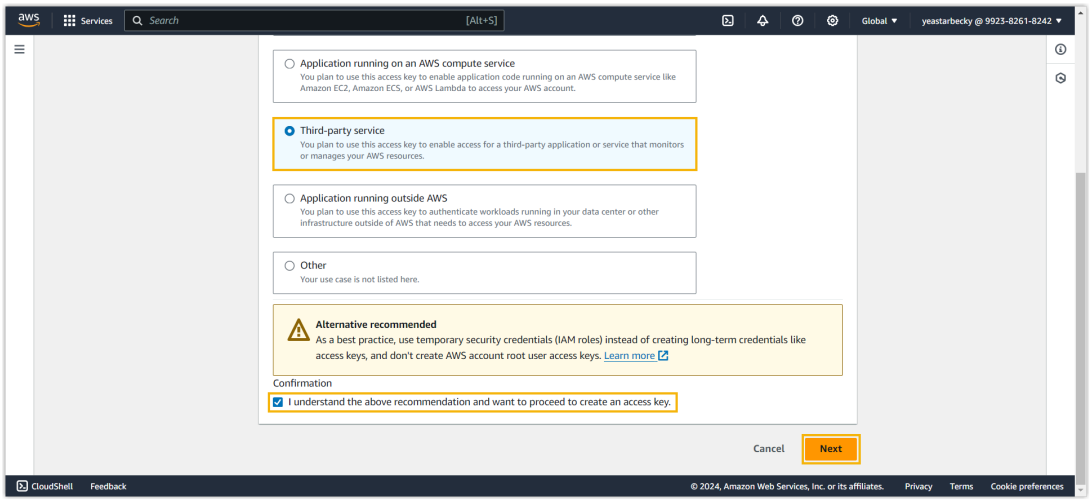
4. Create an access key for the IAM user.
   a. On the users list, click on the newly created IAM user to enter the details page.



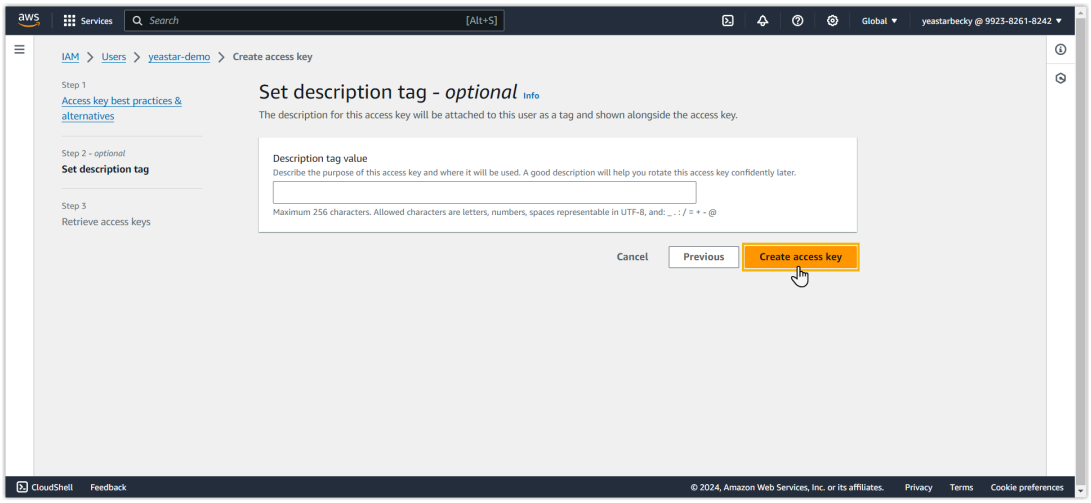
   b. In the **Summary** section, click **Create access key**.



c. On the **Access key best practices & alternatives** page, select **Third-party service**, confirm your operation, then click **Next**.

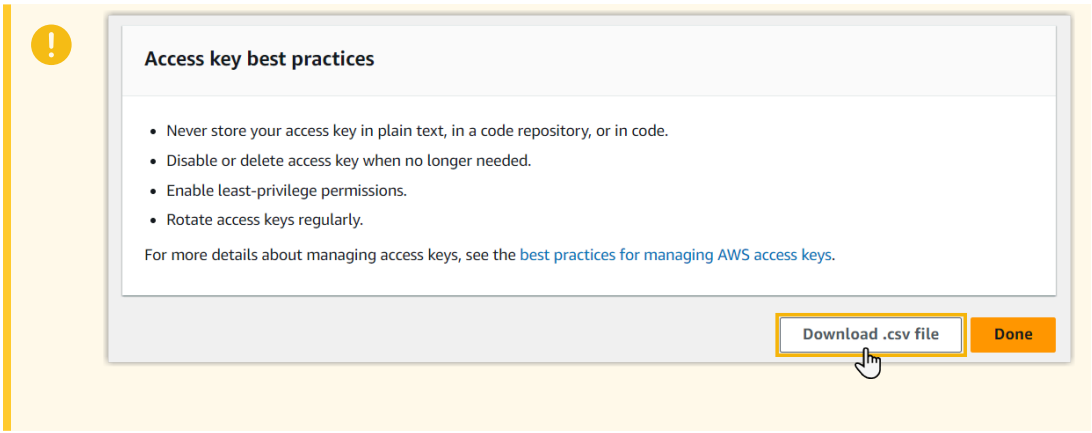


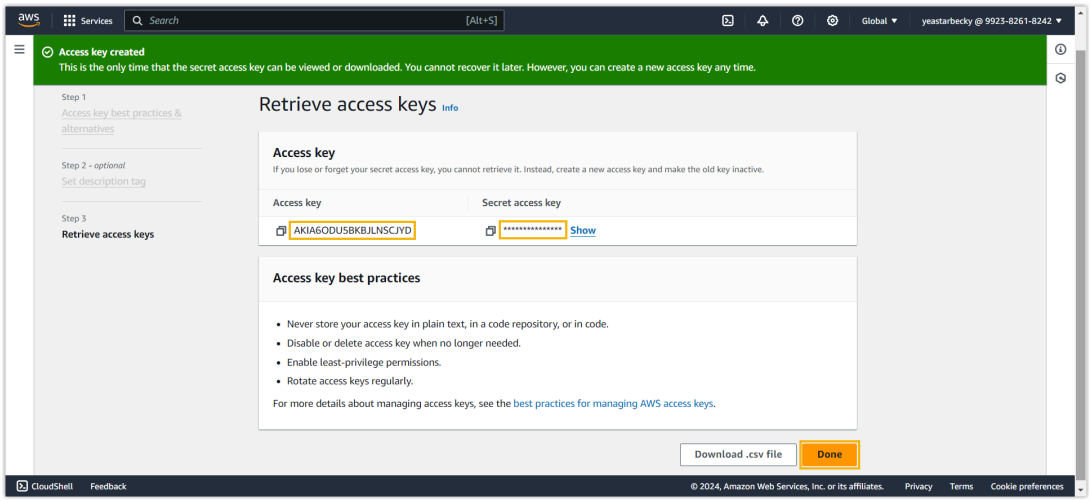
d. On the **Set description tag - optional** page, enter a description for the access key as needed, then click **Create access key**.



e. In the **Retrieve access keys** page, copy and note down the values of **Access key** and **Secret access key**, then click **Done**.

> **Important:**
> For **Secret access key**, the value is shown only ONCE, so make sure that you save the credential in a secure location before clicking **Done** to close the window. We recommend that you click **Download .csv file** to save the credential file to your computer.

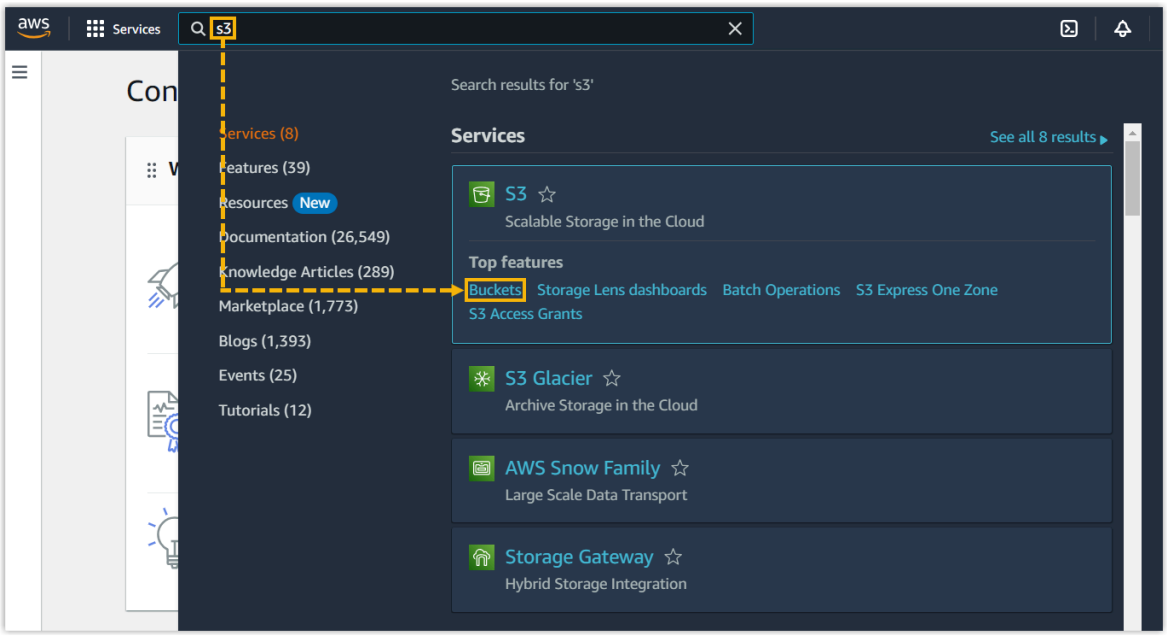




The access key is created and displayed on the access key list.

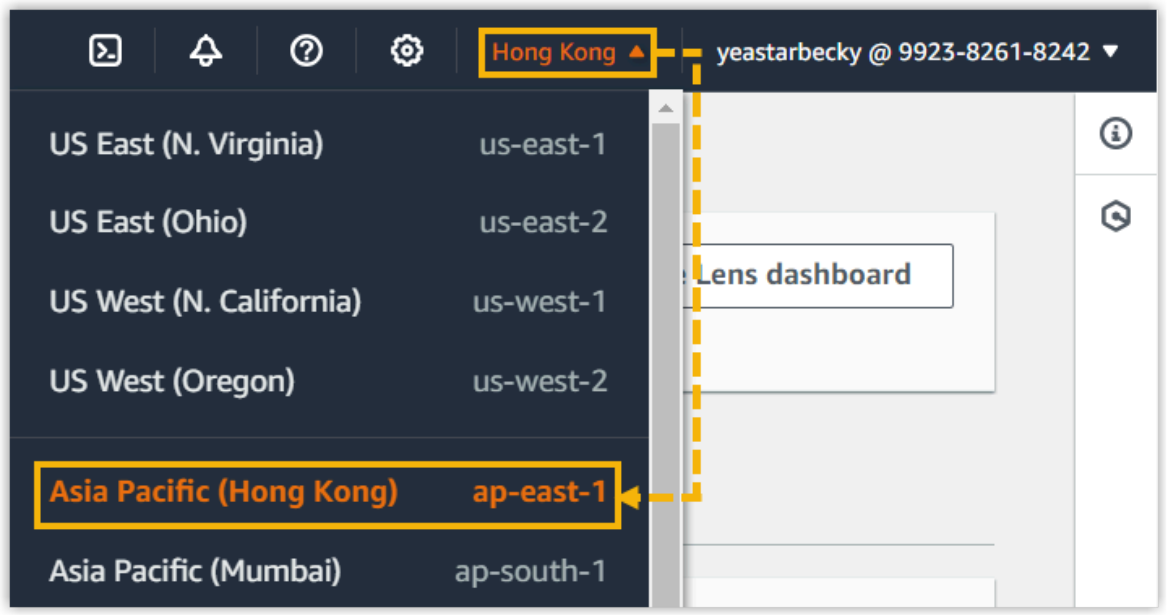
Step 2. Create an Amazon S3 bucket on AWS

1. On AWS Management Console, search for S3 service in the top-left search bar, then select **Buckets**.

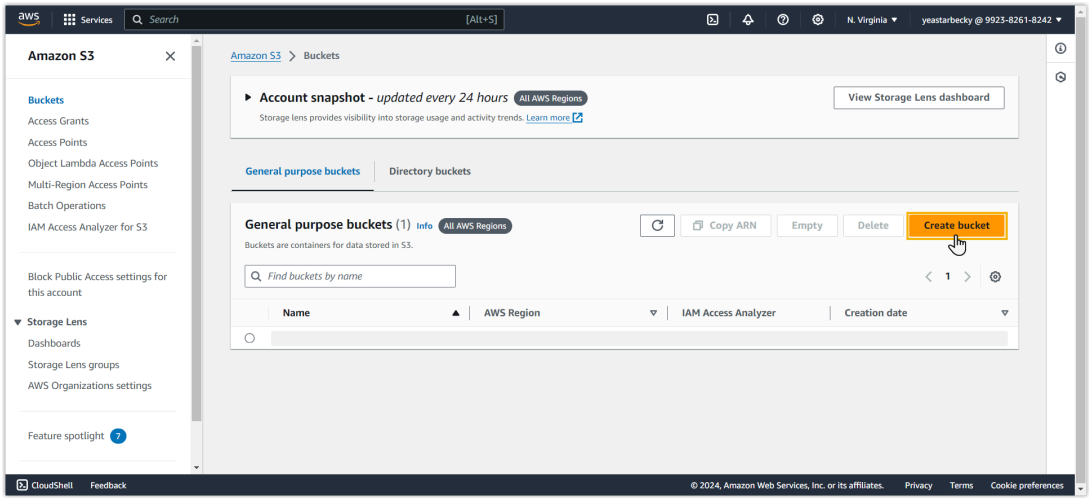


2. At the top-right corner, select the region in which you want to create an S3 bucket.

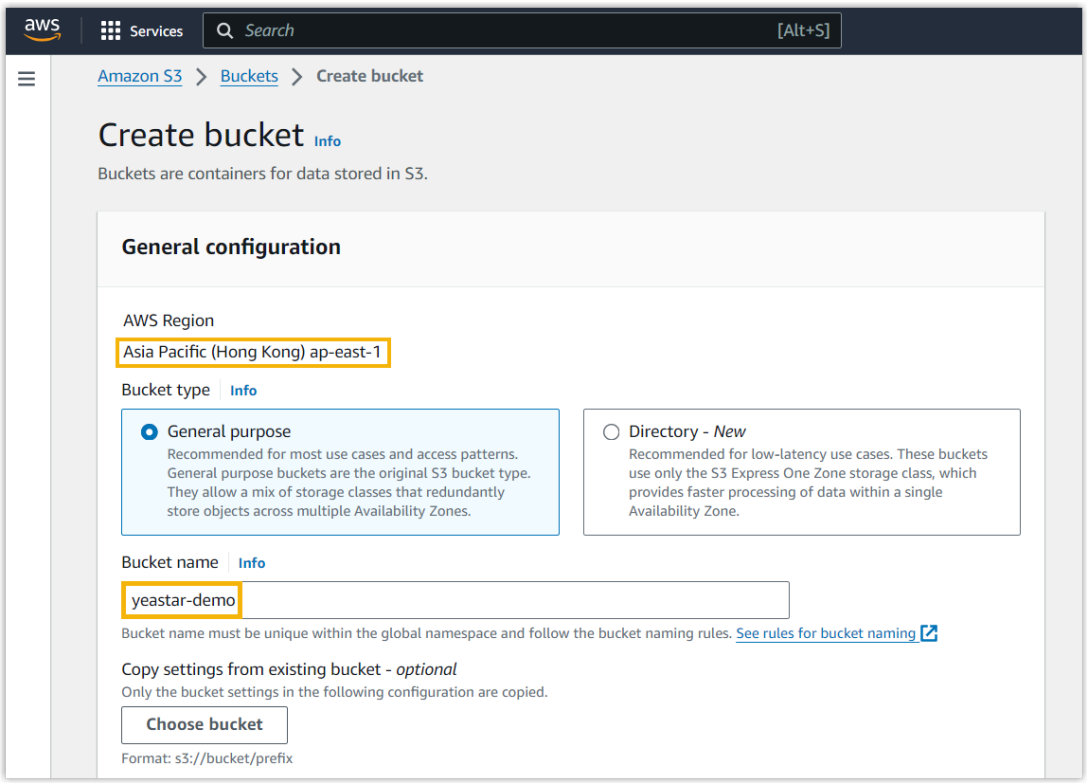
> **Note:**
> To minimize latency and costs and address regulatory requirements, choose a region that is closest to your PBX server.



3. Create an S3 bucket.
    a. On the right of the page, click **Create bucket**.



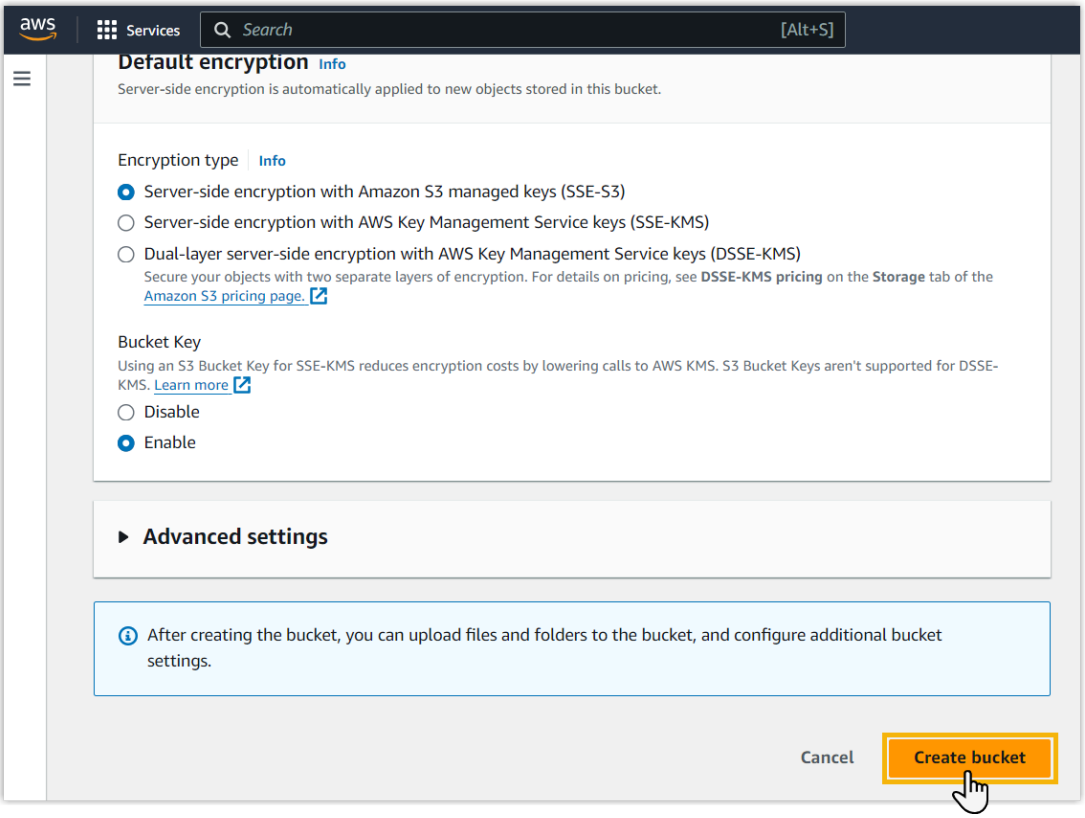
b. Check the region where the bucket will be created and specify a name for the bucket.



- **AWS Region**: Automatically show the region that you have selected.
- **Bucket name**: Enter a name to help you identify the bucket.

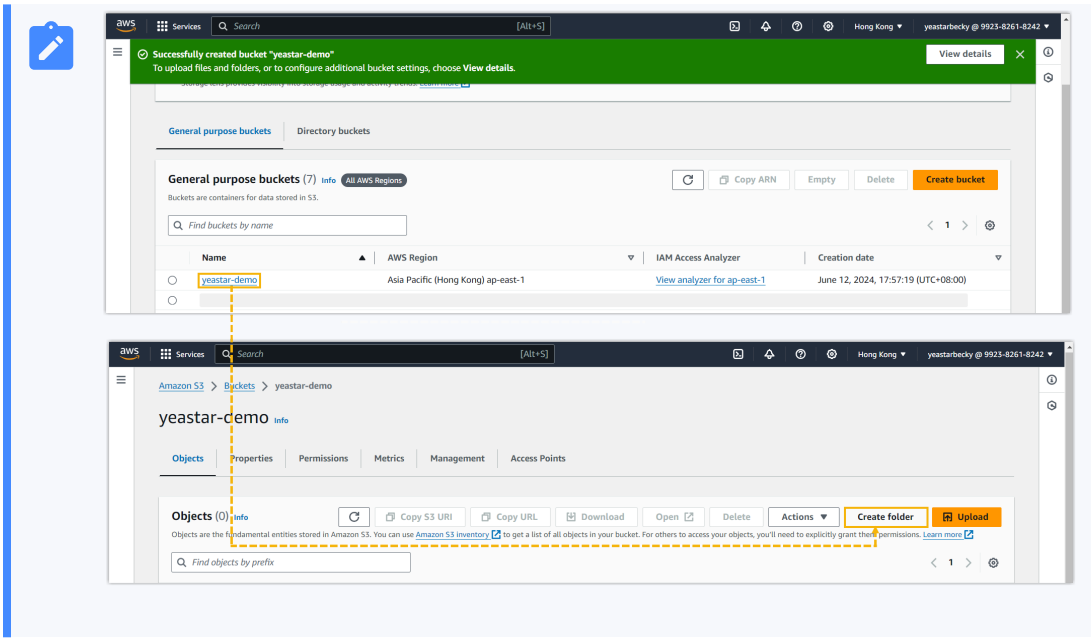
c. Configure other settings according to your needs.

d. At the bottom of the page, click **Create bucket**.



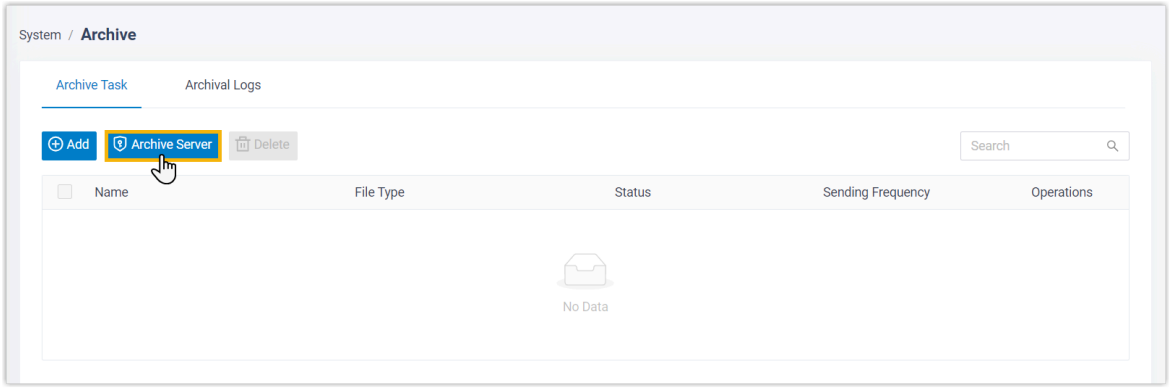
The S3 bucket is created and displayed on the buckets list.

Note:
If you want to group the archived PBX files in Amazon S3, you can create folders in the bucket. In this way, you can specify PBX files to be archived to the designated folder in the follow-up settings.

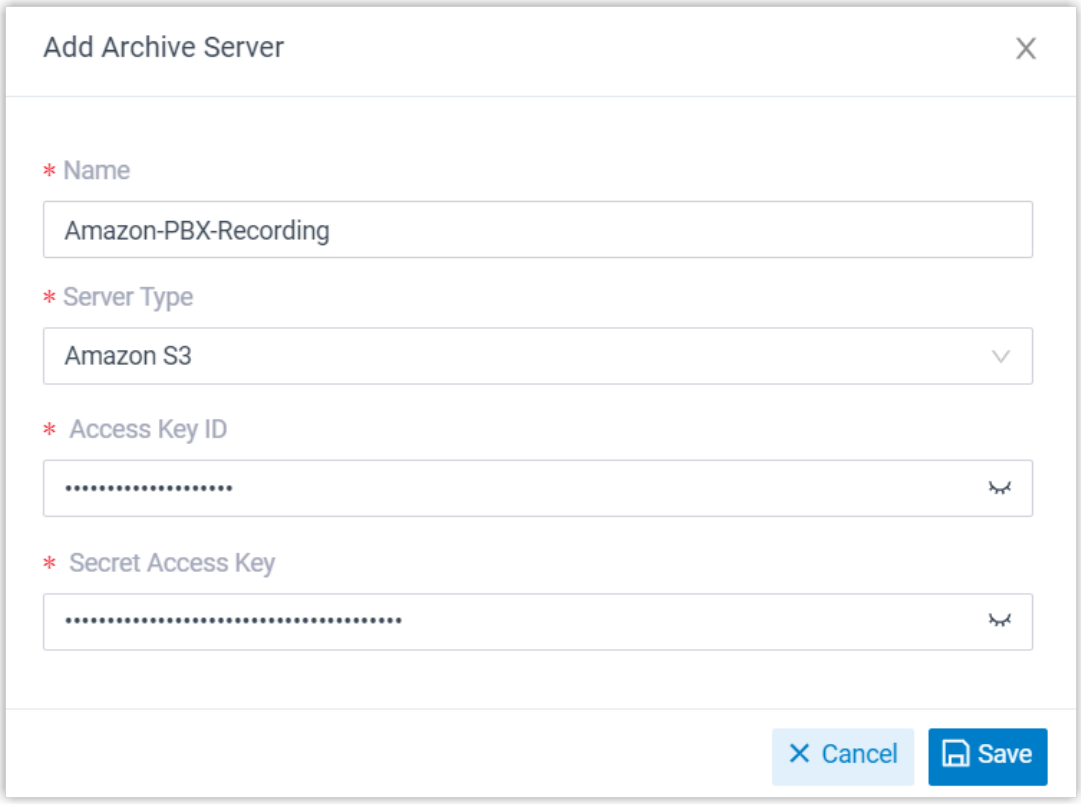


Step 3. Add Amazon S3 bucket as archive server on PBX

1. Log in to PBX web portal, go to **System > Archive**.
2. Under the **Archive Task** tab, click **Archive Server**.



3. Set up Amazon S3 bucket as an archive server.
    a. Click **Add**.
    b. In the pop-up window, complete the following settings.



Setting	Description
Name	Enter a name to help you identify the server.
Server Type	Select **Amazon S3**.
Access Key ID	Enter the access key that you have obtained from AWS.
Secret Access Key	Enter the secret access key that you have obtained from AWS.

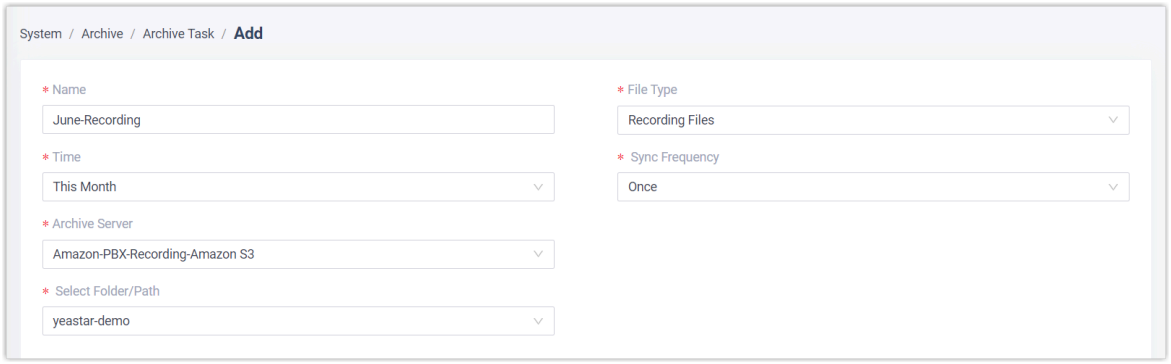
c. Click **Save**.

The Amazon S3 bucket is added as an archive server and displayed on the archive server list.

4. Click ✕ to close the window.

Step 4. Create a task to archive files to Amazon S3 on PBX

1. Under **Archive Task** tab, click **Add**.
2. Create a one-time or recurring archive task.



Setting	Description
Name	Enter a name to help you identify the task.
File Type	Select **Recording Files** or **Backup Files** as needed.
Time	Specify a time range of the files to be archived. **Note:** You can archive files for up to 31 days at a time.
Sync Frequency	Set how often to archive files to Amazon S3. **Note:** As large amounts of data will consume PBX's CPU resources, we recommend that you schedule archive tasks during off-peak hours. • **Once**: If you choose the option, the system will archive files immediately after you save the task. • **Daily**: If you choose the option, select a time from the drop-down list. The system will archive files at this time of the day. • **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system will archive files at this time of the week. • **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system will archive files on this day and time of the month.
Archive Server	Select the Amazon S3 bucket that you have added.
Select Folder/Path	Select the bucket or the folder in which you want to store archived files.

3. Click **Save**.

Result

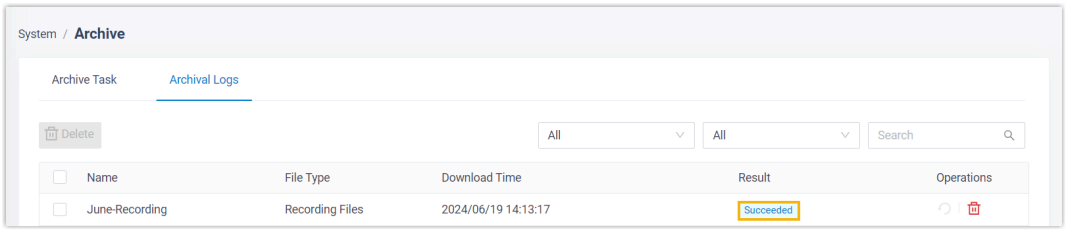
The specified files will be archived to the designated bucket or folder in Amazon S3 immediately or at the scheduled time.

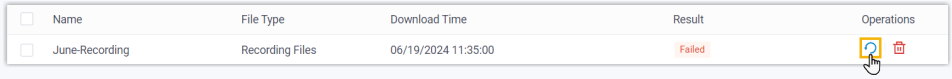
> **Note:**
> The system executes only one task at a time to avoid affecting system performance. If there are multiple tasks, they will be queued up one after another.

You can check the archive result in the following ways.

Check the archive result on PBX

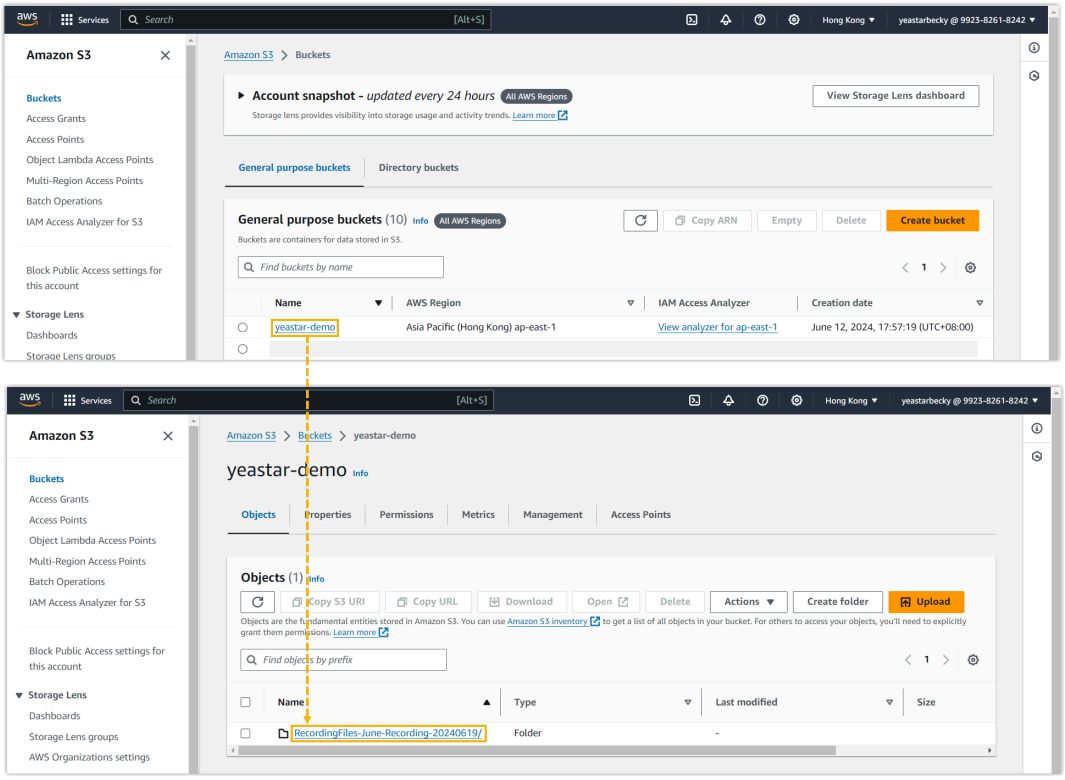
On PBX web portal, go to **System > Archive > Archival Logs**. If the **Result** column of the task shows **Succeeded**, it indicates that the specified files have been successfully archived to Amazon S3.



> **Note:**
> If the task is failed, the **Failed to Archive File(s)** event will be triggered; You can click to retry the task.
>
> 
>

Check the archive result on AWS

On AWS Management Console, go to the created S3 bucket. If the specified files appear in the list, it indicates that the archive is successful.



Archive Files to Google Cloud Storage

Yeastar P-Series Cloud Edition supports archiving the system's call recordings and backup files to Google Cloud Storage, either on a regular interval or at any time you want. This topic describes how to add Google Cloud Storage bucket as an archive server and schedule tasks to archive the desired files.

Requirements and restrictions

Requirements

- **PBX Firmware**: Version 84.15.0.22 or later
- **PBX Plan**: Enterprise Plan (for self-managed phone system) or Ultimate Plan (for Yeastar hosted phone system)

Restrictions

- **Archive server**: 10
- **Archive task**: 200

Introduction

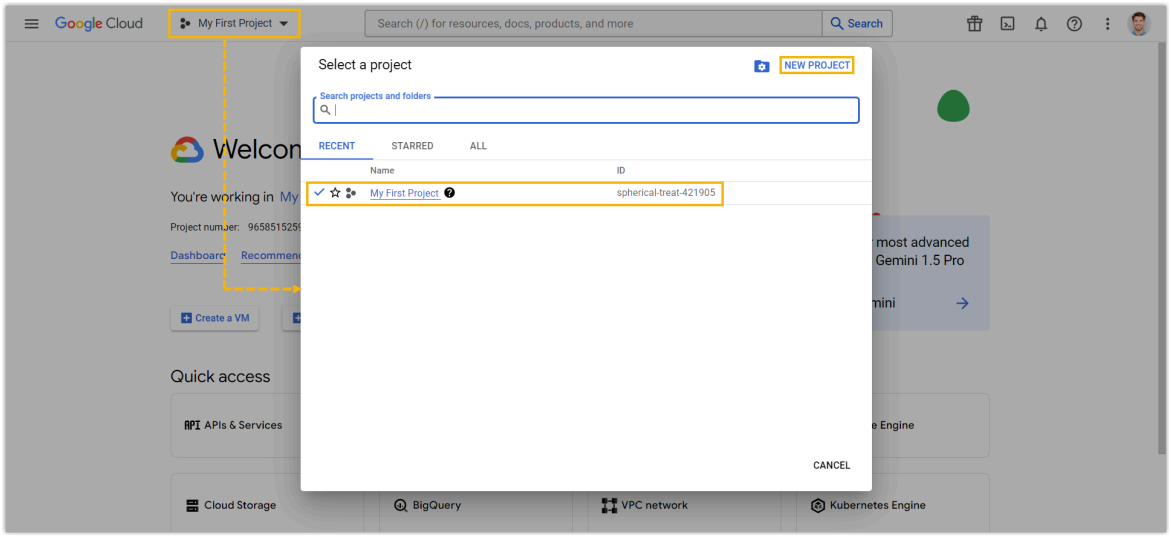
To archive PBX's call recordings and backup files to a specific bucket in Google Cloud Storage, you need to complete the followings:

1. Prepare an eligible account and bucket on Google Cloud.
    - A **service account** that meets the requirements listed below:
        - Full access to Google Cloud Storage resources to archive PBX files
        - An access key to authenticate file archiving requests from PBX
    - A **bucket** to store archived PBX files
2. Set up archive server and task on PBX.
    - Add the Google Cloud Storage bucket as an archive server
    - Create tasks to archive PBX files to Google Cloud Storage

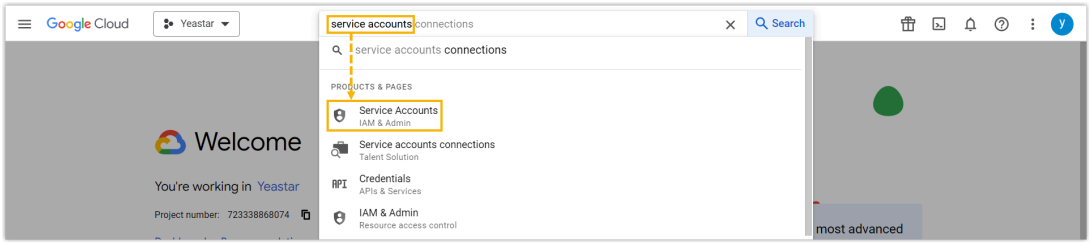
When it is time to execute the archive task, PBX will use the access key of the service account to send file archiving requests to Google Cloud Storage. If the requests are considered authentic, the specified files will be archived to the designated bucket in Google Cloud Storage.

Step 1. Create a service account on Google Cloud

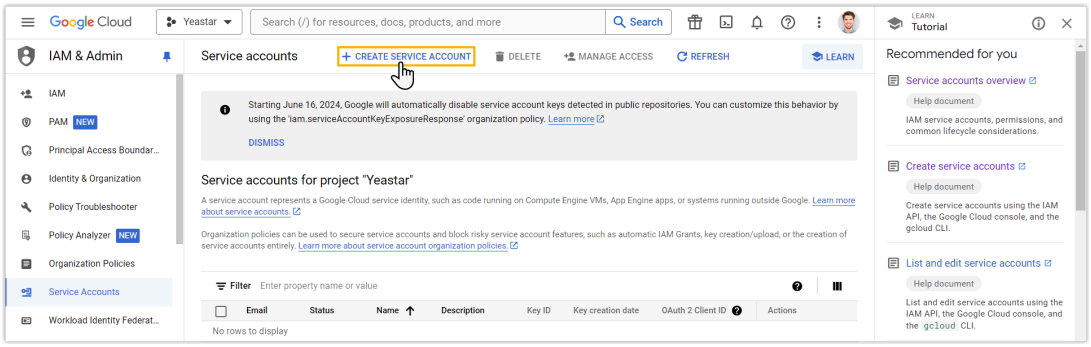
1. Log in to Google Cloud Console.
2. On the project selector page, select or create a Google Cloud project.



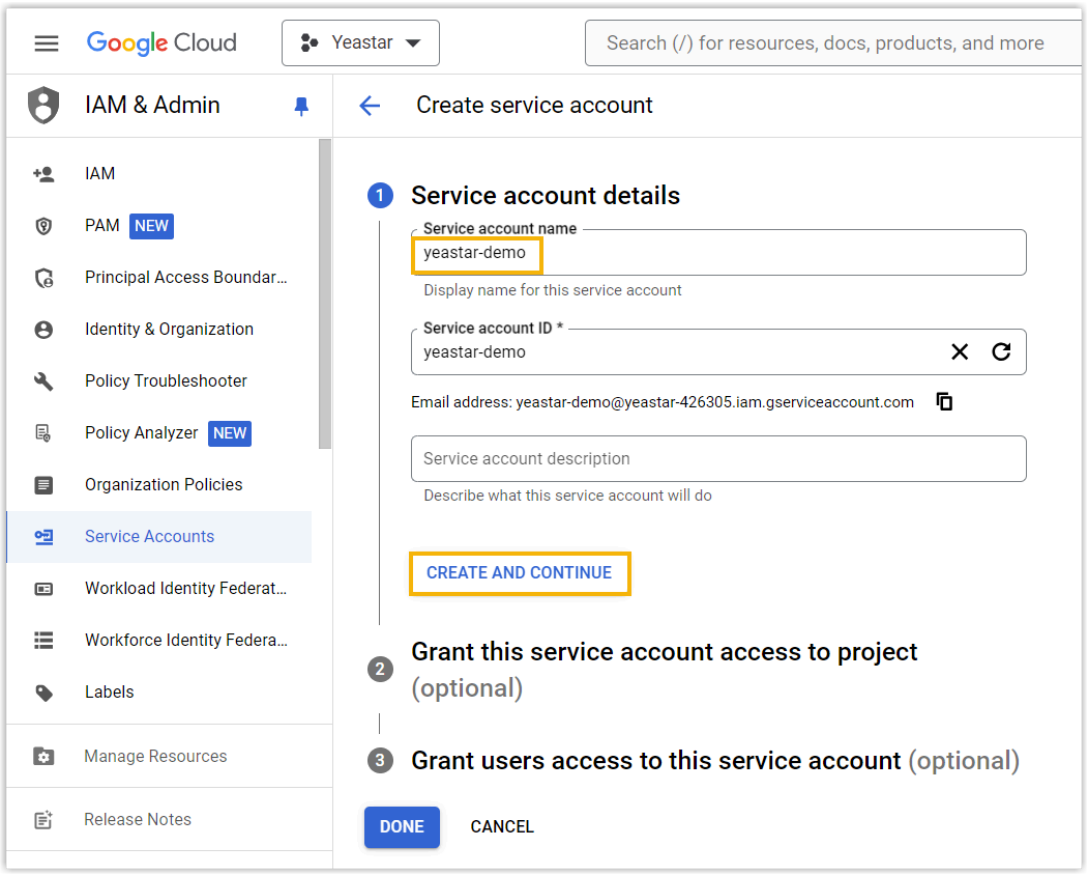
   In this example, we create a new project "Yeastar".
3. Create a service account.
    a. In the top search bar, search for and select **Service Accounts**.



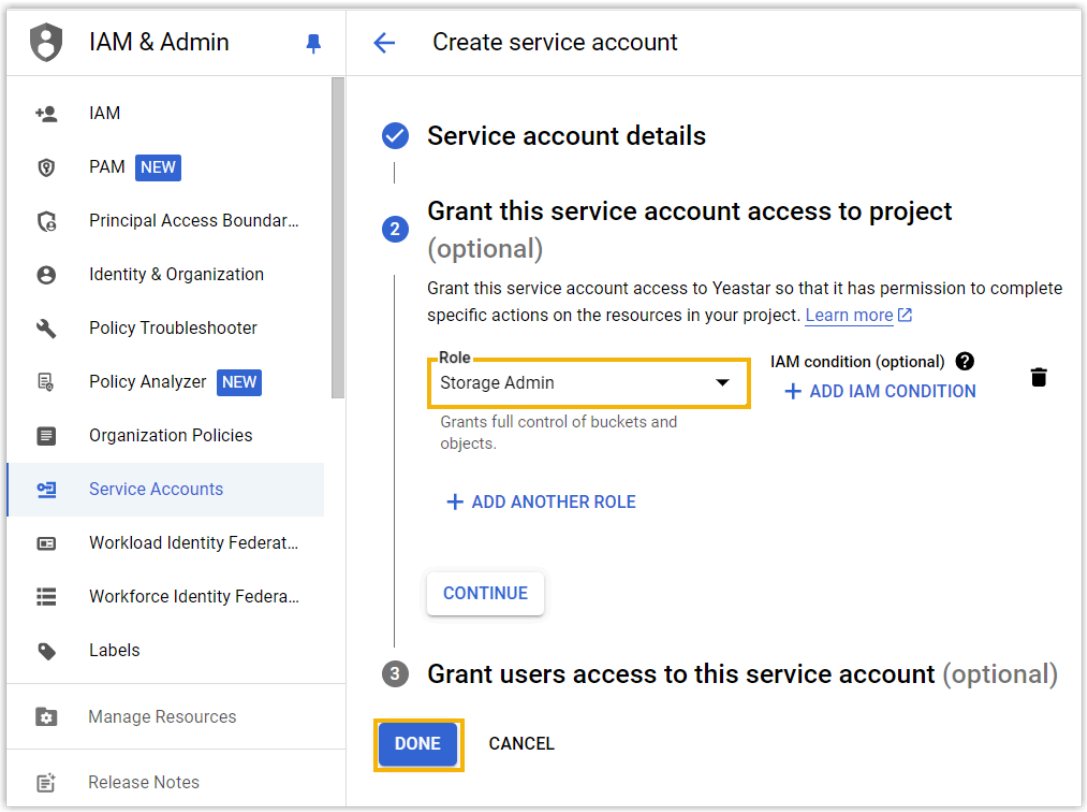
b. In the top pane, click **CREATE SERVICE ACCOUNT**.



c. In the **Service account details** section, enter a name in the **Service account name** field to help you identify the account, then click **CREATE AND CONTINUE**.



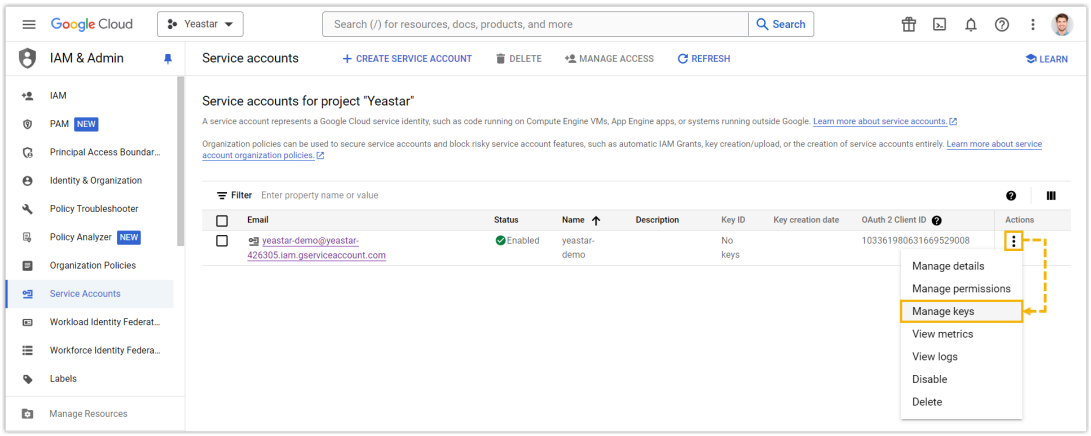
d. In the **Grant this service account access to project (optional)** section, search and select **Storage Admin** from the **Role** drop-down list, then click **DONE**.



The service account is created and displayed on the service accounts list.

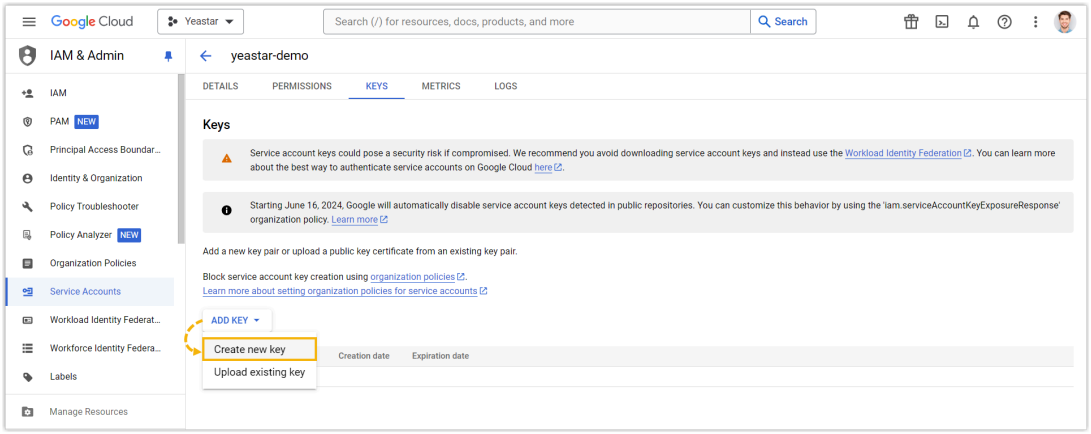
4. Create an access key for the service account.

   a. On the service accounts list, click ⋮ beside the service account that you have created, then select **Manage keys**.

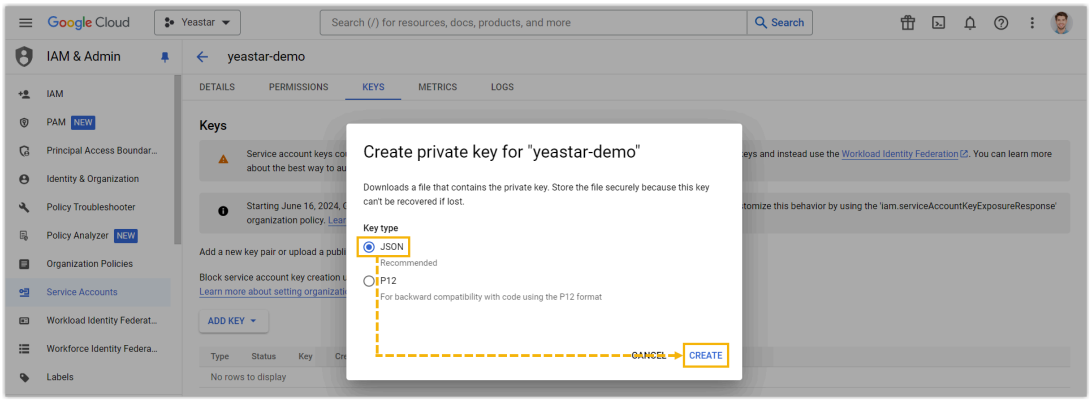


   You are redirected to the key configuration page.

   b. Click **ADD KEY**, then select **Create new key** from the drop-down list.



c. Set **Key type** to **JSON**, then click **CREATE**.



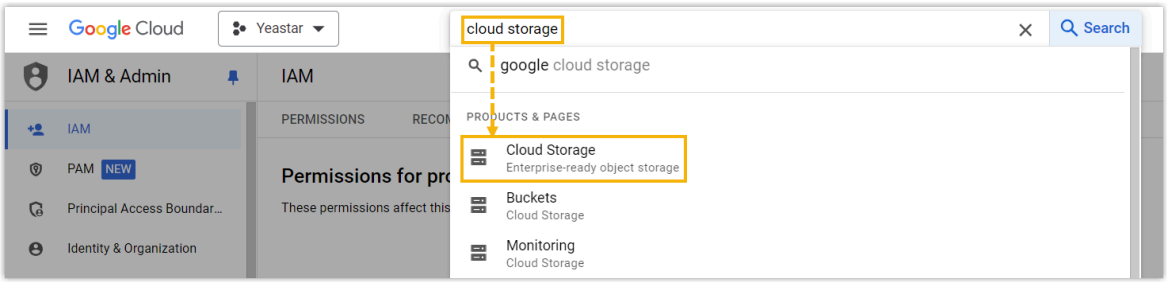
The access key is created and automatically downloaded to your computer as a JSON file.

> **Note:**
> Save the JSON file in a secure location, as you will need the file when setting up Google Cloud Storage bucket as archive server on PBX.

d. Click **CLOSE** to close the pop-up window.

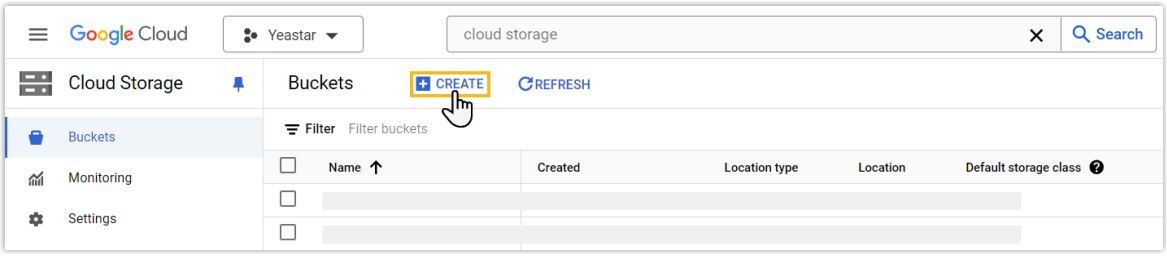
Step 2. Create a bucket on Google Cloud Storage

1. In the top search bar, search for and select **Cloud Storage**.

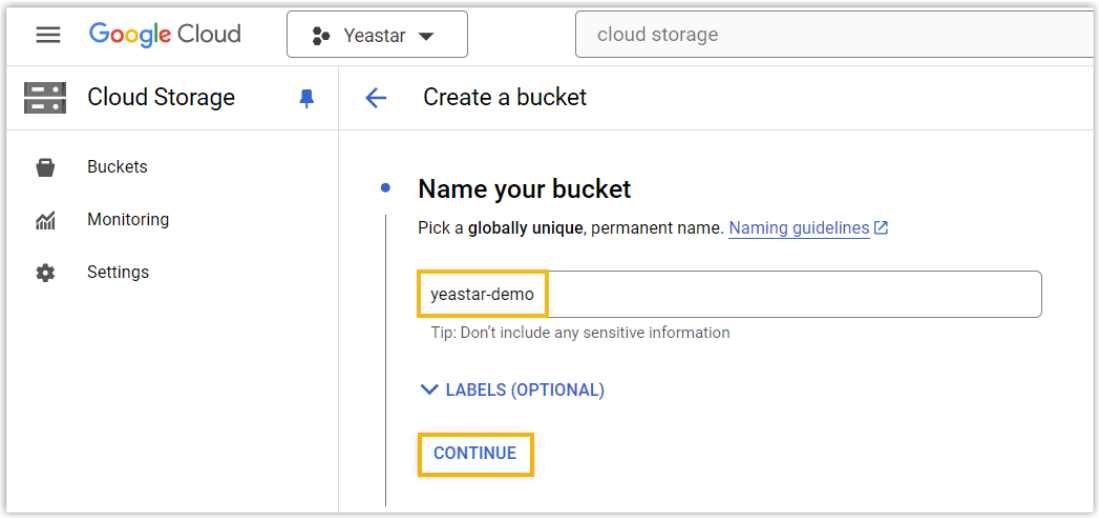


You are redirected to buckets configuration page.

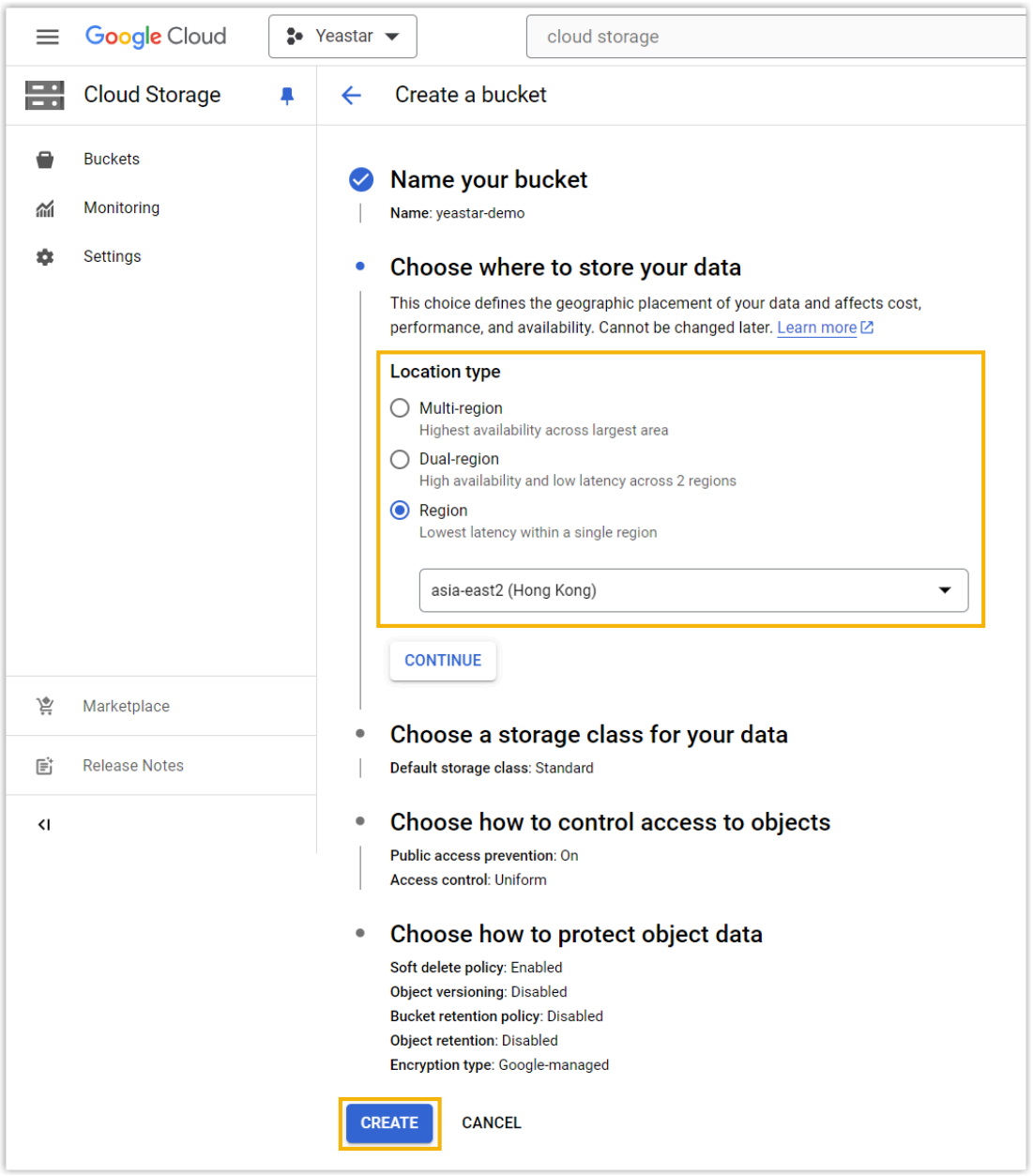
2. In the top pane, click **CREATE**.



3. In the **Name your bucket** field, enter a name to help you identify the bucket, then click **CONTINUE**.

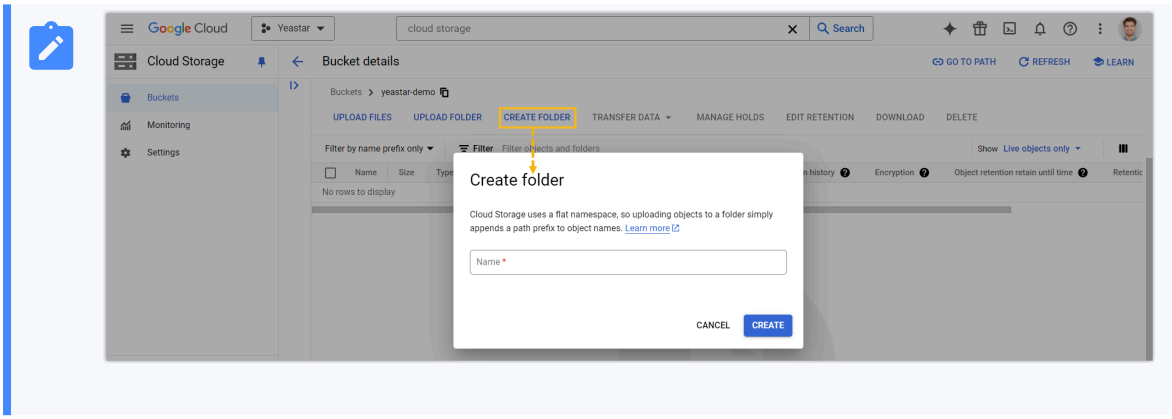


4. In the **Choose where to store your data** section, select the region in which you want to create a bucket, then click **CREATE**.



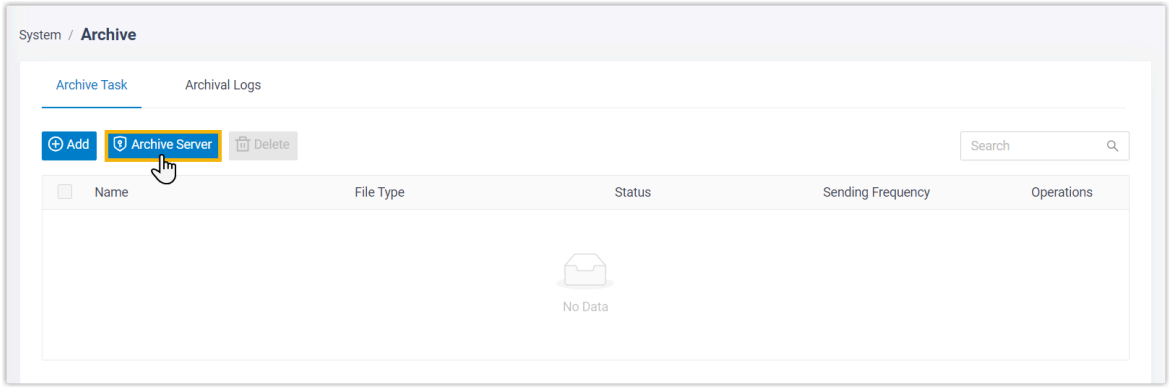
The bucket is created.

Note:
If you want to group the archived PBX files in Google Cloud Storage, you can create folders in the bucket. In this way, you can specify PBX files to be archived to the designated folder in the follow-up settings.

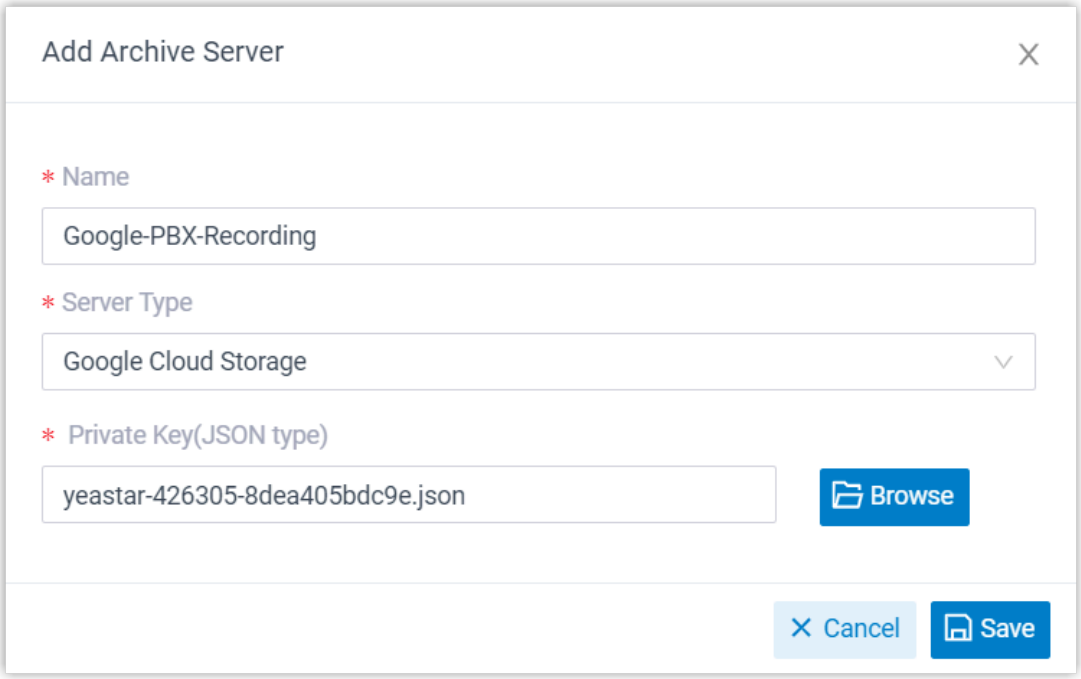


Step 3. Add Google Cloud Storage bucket as archive server on PBX

1. Log in to PBX web portal, go to **System > Archive**.
2. Under the **Archive Task** tab, click **Archive Server**.



3. Set up Google Cloud Storage bucket as an archive server.
    a. Click **Add**.
    b. In the pop-up window, complete the following settings.



Setting	Description
Name	Enter a name to help you identify the server.
Server Type	Select **Google Cloud Storage**.
Private Key(JSON type)	Click **Browse** to browse and upload the JSON file that you have downloaded in Google Cloud.

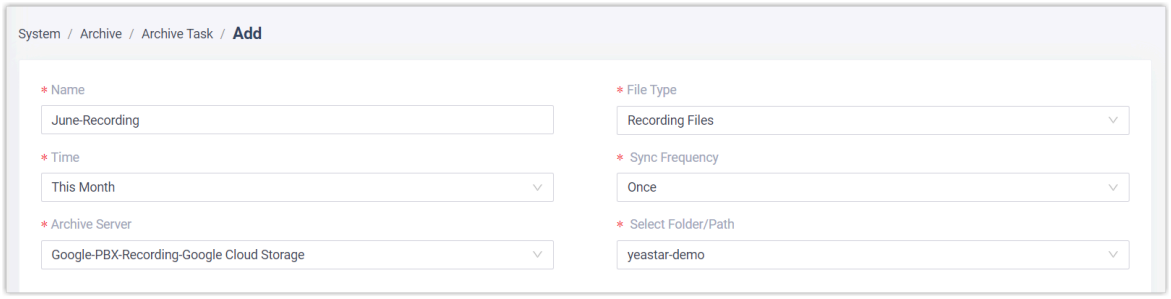
c. Click **Save**.

The Google Cloud Storage bucket is added as an archive server and displayed on the archive server list.

4. Click ✕ to close the window.

Step 4. Create a task to archive files to Google Cloud Storage on PBX

1. Under **Archive Task** tab, click **Add**.
2. Create a one-time or recurring archive task.



Setting	Description
Name	Enter a name to help you identify the task.
File Type	Select **Recording Files** or **Backup Files** as needed.
Time	Specify a time range of the files to be archived. **Note:** You can archive files for up to 31 days at a time.
Sync Frequency	Set how often to archive files to Google Cloud Storage. **Note:** As large amounts of data will consume PBX's CPU resources, we recommend that you schedule archive tasks during off-peak hours. • **Once**: If you choose the option, the system will archive files immediately after you save the task. • **Daily**: If you choose the option, select a time from the drop-down list. The system will archive files at this time of the day. • **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system will archive files at this time of the week. • **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system will archive files on this day and time of the month.
Archive Server	Select the Google Cloud Storage bucket that you have added.
Select Folder/Path	Select the bucket or the folder in which you want to store archived files.

3. Click **Save**.

Result

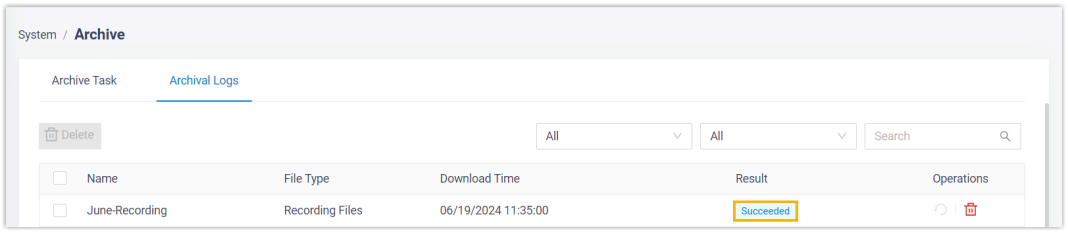
The specified files will be archived to the designated bucket or folder in your Google Cloud Storage immediately or at the scheduled time.

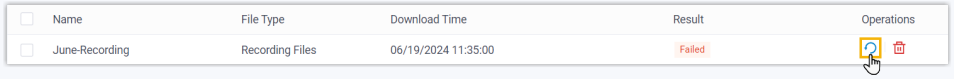
> **Note:**
> The system executes only one task at a time to avoid affecting system performance. If there are multiple tasks, they will be queued up one after another.

You can check the archive result in the following ways.

Check the archive result on PBX

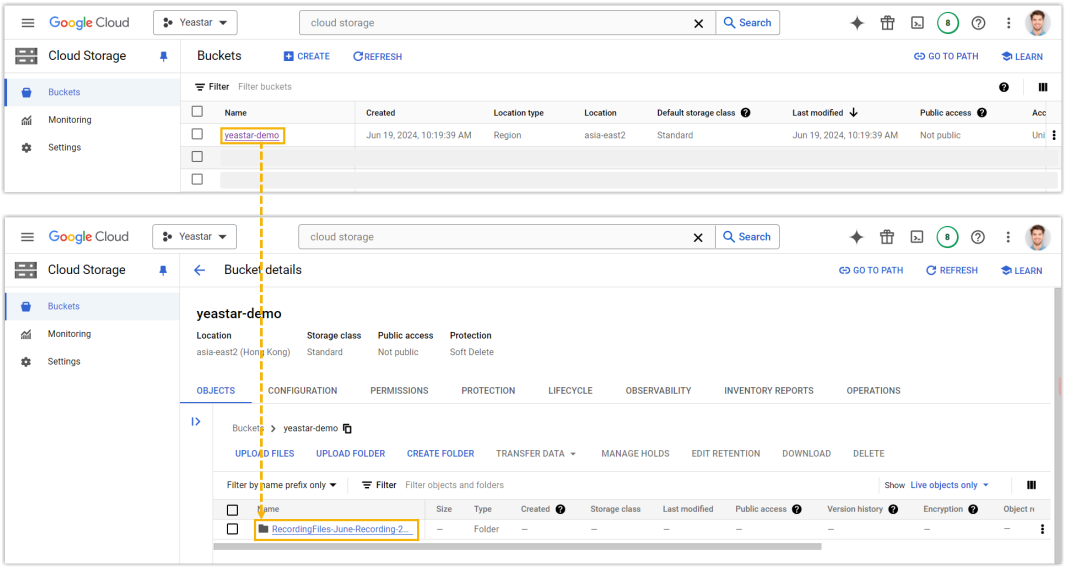
On PBX web portal, go to **System > Archive > Archival Logs**. If the **Result** column shows **Succeeded**, it indicates that the specified files have been successfully archived to Google Cloud Storage.



> **Note:**
> If the task is failed, the **Failed to Archive File(s)** event will be triggered; You can click to retry the task.
>
> 
>

Check the archive result on Google Cloud Storage

On Google Cloud Console, go to the created bucket. If the specified files appear in the list, it indicates that the archive is successful.



Event Notification

Event Notification Overview

Event Notification feature is designed to provide information about changes on Yeastar P-Series Cloud Edition and helps you monitor operations on the PBX. When an event occurs, the system will record the event and notify contacts concerned via specific methods. This topic describes event types, event levels, notification methods, notification email templates, and auto cleanup of events.

Event types

Yeastar P-Series Cloud Edition supports the following event types:

- **Operations**
- **Telephony**
- **System**
- **Security**
- **Reminder**

Table 43. Operations

Event	Description
Administrator Login Success	The administrator successfully logged in to the PBX management portal.
Web User Login Success	A user successfully logged in to the PBX web portal or the Linkus Web Client.
Web User Login Failed	A user failed to log in to PBX web portal or the Linkus Web Client.
Linkus Client Login Failed	An extension user failed to log in to Linkus Mobile Client or Linkus Desktop Client.
Administrator Password Changed	The administrator's password was changed.
Extension User Password Changed	An extension user's user password was changed.
RPS Request Success	The RPS request of the IP Phone(s) succeeded.
RPS Request Failed	The RPS request of the IP Phone(s) failed.

Table 44. Telephony

Event	Description
SIP Trunk Registration Failed	Failed to register or connect to a SIP trunk.
SIP Trunk Re-registered	Successfully re-registered or re-connected to a SIP trunk.
Emergency Call Dialed Out	An extension user placed an emergency call.

Table 45. System

Event	Description
Auto Cleanup Reminder	Reach 90% of the allowed storage limit.
System Reboot	Either of the following situations triggered the event: • The PBX rebooted after configuration. • The PBX automatically rebooted after system crash.
System Restore	The PBX was restored.
New System Firmware Detected	The PBX automatically detected a new firmware version.
System Upgrade Completed	The PBX was upgraded.
Default SMTP Server Error	Default SMTP server failed to send emails. **Note:**

Table 45. System (continued)

Event	Description
	When this event occurs, the system will only notify the contacts via the notification methods of **Call Extension** or **Call Mobile**. Please make sure these notification methods have been enabled in notification contacts.
Recording Usage is About to Reach the Capacity Limit	The recording is about to reach the total capacity limitation.
Abnormal Core Call Services	A deadlock issue has been detected in the core call process.
Core Call Services Recovery Completed	The core call process has recovered successfully.

Table 46. Security

Event	Description
Web User Locked Out	PBX blocked the source IP when either of the following situations was met: • Web Login failure for more than 5 times in 24 hours. • More than 5 accounts were locked in 24 hours.
Linkus User Blocked Out	PBX blocked the source IP when either of the following situations was met: • Login failure (Linkus Mobile Client or Linkus Desktop Client) for more than 5 times in 24 hours. • More than 5 accounts were locked in 24 hours.
Extension Registration Blocked Out	PBX blocked the source IP when either of the following situations was met: • Registration failure for more than 20 times. • More than 3 accounts were locked.
Auto Defense IP Blocked Out	The monitored service or port reached the limit of **Number of Packets** during specific **Time Interval**.
Outbound Call Frequency Exceeded	An extension has exceeded the limit of **Number of Calls** during specified **Time Period** set in an Outbound Call Frequency Restriction rule.
Outbound Call to a Disallowed Country	An extension user made an outbound call to a disallowed country.
API Authentication Blocked Out	PBX blocked the source IP due to too many failed API authentication attempts.

Table 46. Security (continued)

Event	Description
Linkus SDK Authentication Blocked Out	PBX blocked the source IP due to too many failed Linkus SDK authentication attempts.

Table 47. Reminder

Event	Description
Plan Expiration Reminder	The current plan will expire soon.
Video Conferencing Usage Has Reached 90% of Time Limit	Reach 90% of the annual time limit of video conferencing.
Video Conferencing Usage Limit Reached	Reach annual usage limit of video conferencing.
Failed to Archive File(s)	The task to archive recording files or backup files to an external server has failed.

Event levels

Event level is used to indicate how severe or important an event is. Choosing an appropriate level prevents recipients from receiving repetitive information.
Yeastar P-Series Cloud Edition supports the following event levels:

- **Information**: Events that pass general information to recipients.
- **Warning**: Events that indicate specific components or applications are not in ideal states, and further action could result in errors.
- **Alert**: Events that indicate problems require timely attention.

> **Note:**
> - When an event occurs, the system gives you a pop-up reminder on the right of PBX web portal.
> - For event whose default level is not **Alert**, the system will NOT give you a pop-up reminder even if you change the level from **Information** or **Warning** to **Alert**.

Notification contacts and methods

You can set notification contacts to internal users or external users, and notify users in the following ways when events occur:

- **Send Email**
- **Call Extension**
- **Call Mobile**

For more information, see Manage Notification Contacts.

Notification email templates

If notification method is set to **Send Email** for a specific contact, the system will send notification emails in corresponding email template when an event occurs. Yeastar P-Series Cloud Edition provides default email template for each event, you can also customize email templates according to your needs.

For more information, see Customize Email Templates.

Auto cleanup of event logs

By default, when event logs reach 50,000, the system automatically deletes the oldest logs. You can change the value, or set the maximum days that logs can be retained.

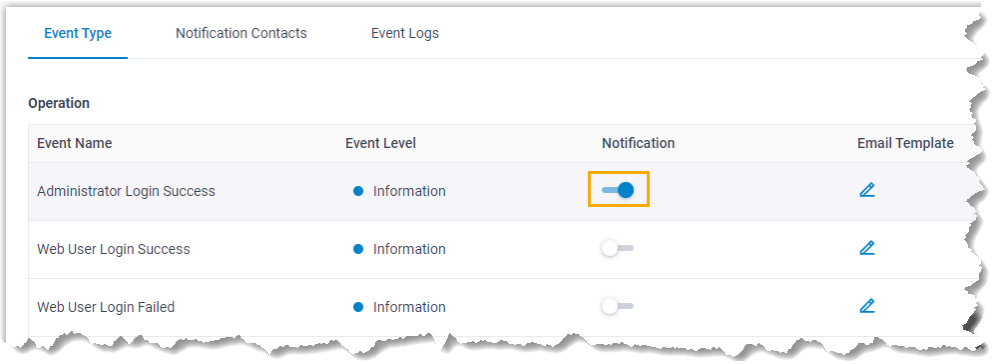
For more information, see Auto Cleanup Settings.

Configure Event Notifications

This topic describes how to configure event notifications.

Procedure

1. Log in to PBX web portal, go to **System > Event Notification > Event Type**.
2. In the **Notification** column, enable notifications for desired event.



3. Configure notification settings for a desired event.

- **Event Level**: A proper level helps you identify seriousness of an event. Use default level or select a level from the drop-down list.
- **Email Template**: To customize template of the email that will be sent to relevant contacts when the event occurs, click .
- **Notification Contacts**: Add notification contacts and select proper notification methods.

  For more information, see Manage Notification Contacts.

Result

When the event occurs, the followings can be achieved:

- The PBX sends notifications to relevant contacts via specific notification methods.
- On **Event Trend** section, the event is included in the statistics of corresponding event level.
- At the top right corner of the page, automatically adds 1 in the color that indicates the event level.

> **Note:**
> If default level for the event is **Error**, the system also gives you a pop-up reminder on the right of PBX web portal.

What to do next

At the top right corner, click to check event details.

Manage Notification Contacts

This topic describes how to add, edit, or delete a notification contact.

Add a notification contact

1. Log in to PBX web portal, go to **System > Event Notification > Notification Contacts**, click **Add**.
2. In the pop-up window, configure contact settings.
   - **Notification Contact**: Select an internal user or set an external user. If you choose **Custom**, enter a name in the **Contact Name** field.
   - **Notification Methods**: Set how to notify the contact when events occur.

- **Call Extension**: The PBX will call the extension number of the contact when an event occurs.
- **Send Email**: The PBX will send notifications to the email address of the contact when an event occurs.
- **Call Mobile**: The PBX will call the mobile number of the contact when an event occurs.

> **Note:**
> To ensure that PBX can successfully call the mobile number, make sure that the **Prefix** is configured correctly according to the outbound route rule.

- **The Event Levels to Notify**: Select the level of events that you want to notify the contact. The contact will only receive notifications when events at the level occur.

3. Click **Save**.

Edit a notification contact

1. Log in to PBX web portal, go to **System > Event Notification > Notification Contacts**.
2. Select a desired contact, click .

> **Note:**
> To edit the event notifications of super administrator, click the at the top-right corner and select **Administrator Settings**.

3. Change the notification methods and notification level according to your needs.
4. Click **Save**.

Delete notification contacts

1. Log in to PBX web portal, go to **System > Event Notification > Notification Contacts**.
2. Delete one or more contacts according to your needs.
   - To delete a contact, click beside the desired contact, click **OK**.

- To delete contacts in bulk, select the checkboxes of the desired contacts, click **Delete** and **OK**.

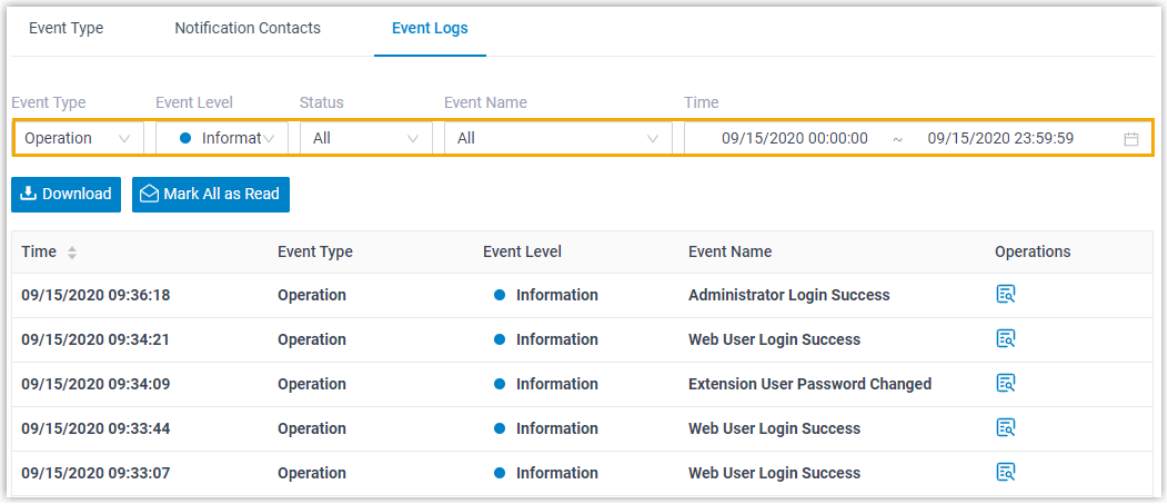
The contacts are removed from the list, and will not receive notifications when events occur.

Manage Event Logs

All the occurred events are saved in event logs so that you can trace the events. This topic describes how to view, download, and mark event logs as read.

Procedure

1. Log in to PBX web portal, go to **System > Event Notification > Event Logs**.
2. Set the search criteria to search events.



- **Event Type**: Search all the event logs or search logs by a specific event type.
- **Event Level**: Search all the event logs or search logs by a specific event level.
- **Status**: Search all the event logs or search logs by a specific acknowledgement status.
- **Event Name**: Search all the event logs or search a specific event.
- **Time**: Set the start date and end date of the events.

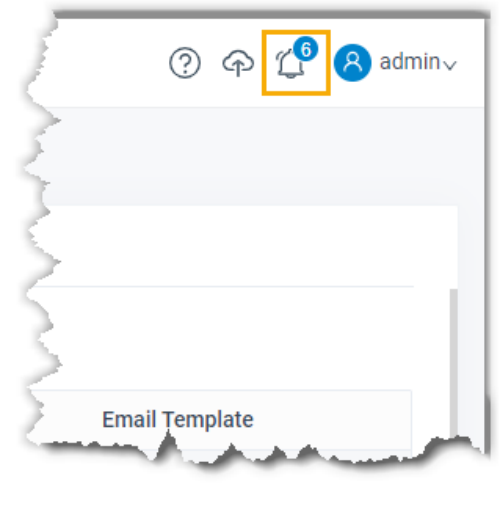
The matched events are displayed on the page.

3. Handle the searched events according to your needs.
   - To check log details, click beside the desired event log.

     The event log will be marked as read.

- To download all the searched logs, click **Download**.
- To mark all the searched logs as read, click **Mark All as Read**.

Note:
At the top right corner, the number of unread events of the event level prompted on will be cleared.



Security

Security Overview

Yeastar P-Series Cloud Edition provides robust security options to ensure a secure and reliable phone service to your business operation.

Allowed IPs

Allowed IPs is used to add specific IP addresses or domain names to the PBX allowlist to prevent from being blocked by the system.

For more information, see Add an Allowed IP Address and Manage Allowed IP Addresses.

Blocked IPs

The blocked IP addresses would be listed in the **Blocked IPs**. If a trusted IP address was blocked, you can go to **Blocked IPs** to delete the IP address.

For more information, see Manage Blocked IP Addresses.

Outbound Call Frequency Restriction

Outbound Call Frequency Restriction rule is used to limit the number of outbound calls over specified time period.

The PBX has a default rule to limit extension users to make maximum 5 outbound calls in 1 second.

You can also set up new rules according to your needs. For more information, see Add an 'Outbound Call Frequency Restriction' Rule.

Console/SSH Access

Yeastar P-Series Cloud Edition supports SSH access. Technical supporter engineers can establish a temporary SSH connection on the PBX to check logs and debug the PBX.

For more information, see Access the System via SSH.

Allowed Country IPs

You can set up **Allowed Country IPs** to only allow specific countries or regions to access your phone system, thus preventing the situations that hackers remotely access your phone

system to make international and long-distance calls, monitor conversations, or do other operations that may cause security threats to your phone system.

For more information, see Restrict Specific Countries or Regions from Accessing Yeastar P-Series Cloud Edition.

Allowed Country Codes

You can set up **Allowed Country Codes** to restrict users from making international calls to specific countries or regions, thus effectively preventing toll fraud.

For more information, see Restrict International Calls to Specific Countries or Regions.

Passwordless Login

If you need the service provider to access your PBX system for remote technical support, and don't want to expose the administrator account credentials, you can set up the PBX to allow passwordless login.

For more information, see Allow Passwordless Login to Yeastar P-Series Cloud Edition.

Allowed IPs

Add an Allowed IP Address

To prevent trusted IP address or domain from being blocked by the system, you can add desired IP addresses or domains to the allowlist.

Procedure

1. Log in to PBX web portal, go to **Security > Security Rules > Allowed IPs**, click **Add**.
2. In the **Name** field, enter a name to help you identify the rule.
3. **Optional:** In the **Description** field, enter the rule description if necessary.
4. In the **Type** drop-down list, select the desired type.
    - **IP Address**: If you choose this type, enter an IP address in the **IP Address** field.
    - **Domain**: If you choose this type, enter a domain in the **Domain** field.
5. Click **Save**.

Result

For IP address or domain that is allowed to access the PBX, the system will always accept connections from the address.

Manage Allowed IP Addresses

This topic describes how to edit or delete the allowed IP address or domain.

Edit an allowed IP address

1. Log in to PBX web portal, go to **Security > Security Rules > Allowed IPs**.
2. Select the desired IP address or domain, click  .
3. Edit the IP address or domain according to your needs.
4. Click **Save**.

Delete allowed IP addresses

1. Log in to PBX web portal, go to **Security > Security Rules > Allowed IPs**.
2. Delete one or more IP addresses or domains according to your needs.
   - To delete an IP address or domain, click  , then click **OK**.
   - To delete IP addresses or domains in bulk, select the checkboxes of the desired IP addresses, click **Delete** and **OK**.

Blocked IPs

Manage Blocked IP Addresses

This topic describes how to view or delete IP addresses that were blocked.

View blocked IP address

1. Log in to PBX web portal, go to **Security > Security Rules > Blocked IPs**.
2. Check details of the IP address that was blocked.
   - **Defense Type**: The defense type.
   - **Block Type**: Whether an account or an IP address was blocked.
   - **Block Range**: The account range or port range that was blocked.

- **Time of Attack**: The time that the blocked account or IP address tried to attack the system.
- **Protocol**: The protocol that the blocked account or IP address tried to attack.
- **Attacked Port**: The port that the blocked account or IP address tried to attack.
- **Source IP Address**: The IP address from which the attack was originated.
- **Expiration Date**: The date and time on which the block would expire.

Delete blocked IP address

1. Log in to PBX web portal, go to **Security > Security Rules > Blocked IPs**.
2. Delete one or more IP addresses according to your needs.
   - To delete an IP address, click  beside the desired IP address, click **OK**.
   - To delete IP addresses in bulk, select the checkboxes of the desired IP addresses, click **Delete** and **OK**.

Outbound Call Frequency Restriciton

Add an 'Outbound Call Frequency Restriction' Rule

For security purpose, we recommended that you use Outbound Call Frequency Restriction rule to restrict the outbound call frequency in Yeastar P-Series Cloud Edition. The PBX has a default rule to limit extension users to make maximum 5 outbound calls in 1 second, you can also set up your own rules according to your need. With the restriction rules, the system can be protected against the threat of toll fraud.

Procedure

1. Log in to PBX web portal, go to **Security > Security Rules > Outbound Call Frequency Restriction**, click **Add**.
2. In the pop-up window, configure the following settings:
   a. In the **Name** field, set a name to help you identify the rule.
   b. Click **Add** and set up the restriction parameters:
      - **Number of Calls**: Set the limit number of outbound calls.
      - **Time Period**: Set a specific time period, and then select the time unit as **Minute(s)** or **Second(s)**.
   c. Click **Save** and **Apply**.

What to do next

Apply the Outbound Call Frequency Restriction rule to limit the extensions. For more information, see Limit Outbound Call Frequency of an Extension.

Manage 'Outbound Call Frequency Restriction' Rules

This topic describes how to edit or delete Outbound Call Frequency Restriction rules.

Edit an 'Outbound Call Frequency Restriction' rule

1. Log in to PBX web portal, go to **Security > Security Rules > Outbound Call Frequency Restriction**.
2. Select the desired rule, click  .
3. Edit rule settings according to your needs.
4. Click **Save** and **Apply**.

Delete 'Outbound Call Frequency Restriction' rules

1. Log in to PBX web portal, go to **Security > Security Rules > Outbound Call Frequency Restriction**.
2. Delete one or more rules according to your needs.
   - To delete a rule, click  beside the desired rule, click **OK**.
   - To delete rules in bulk, select the checkboxes of the desired rules, click **Delete** and **OK**.

Export and Import 'Outbound Call Frequency Restriction' Rules

The Outbound Call Frequency Restriction rules configured in Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired Outbound Call Frequency Restrictions in the exported file, and import the file to PBX again.

Export 'Outbound Call Frequency Restriction' rules

You can export all Outbound Call Frequency Restriction rules to a CSV file, and then make additions, removals, and changes to the file.

1. Log in to PBX web portal, go to **Security > Security Rules > Outbound Call Frequency Restriction**.

2. Click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Outbound Call Frequency Restriction Rule Parameters.

Import 'Outbound Call Frequency Restriction' rules

We recommend that you export Outbound Call Frequency Restriction rules to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the import parameters meet requirements. For more information , see Outbound Call Frequency Restriction Rule Parameters.

Procedure

1. Log in to PBX web portal, go to **Security > Security Rules > Outbound Call Frequency Restriction**.
2. Click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.

   The Outbound Call Frequency Restriction rules in the CSV file will be displayed in the **Outbound Call Frequency Restriction** list.

Related information

Import and Export -FAQ

Console/SSH Access

Access the System via SSH

This topic takes Putty as an example to introduce how to access Yeastar P-Series Cloud Edition via SSH.

Procedure

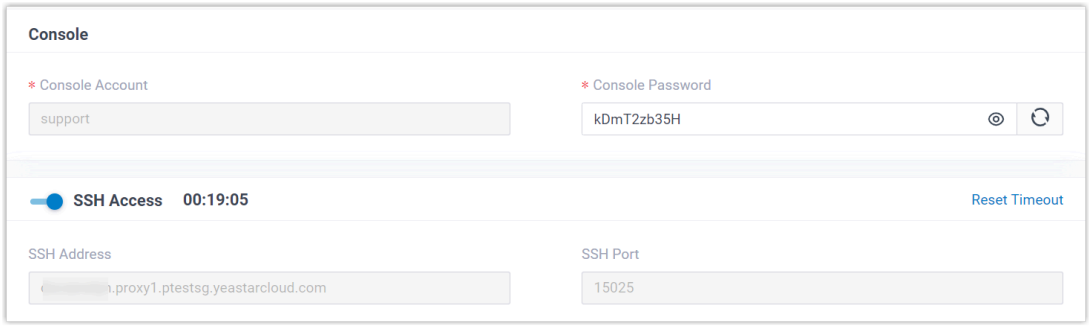
1. Enable SSH, check SSH address, SSH port, console account, and console password on the PBX.
    a. Log in to PBX web portal, go to **Security > Security Settings > Console/SSH Access**.
    b. Enable **SSH Access** and click **OK** in the pop-up window.
    c. Set up the connection timeout in the pop-up window.
        i. In the **Timeout** drop-down list, select the desired time.
        ii. Click **Save**.

    The SSH connection is enabled, and the countdown of connection time is displayed on the web page.

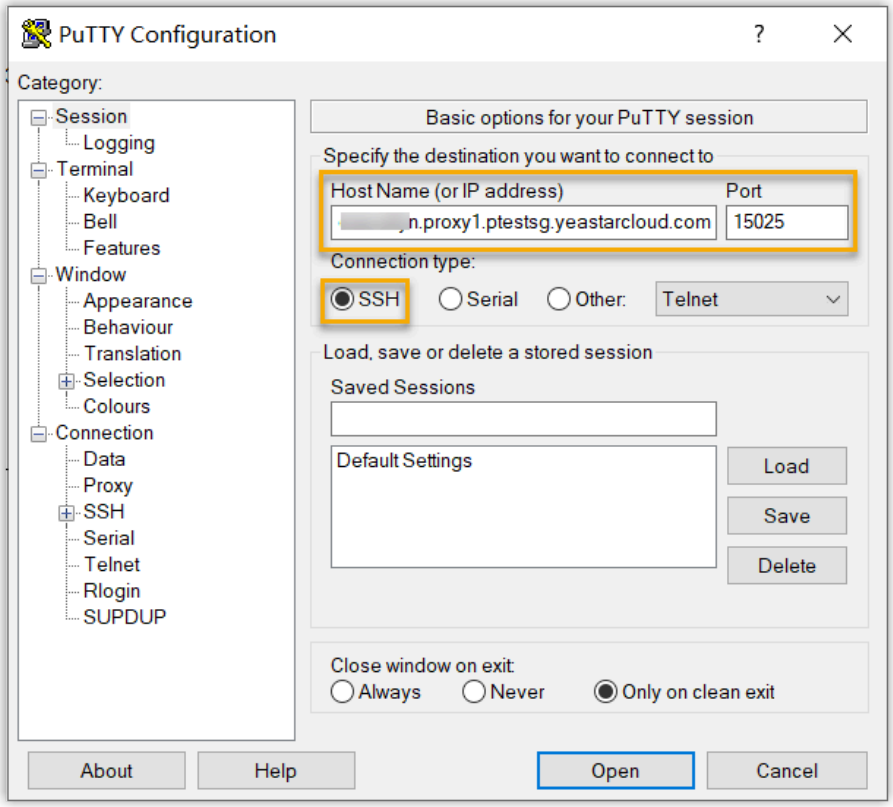
    > **Note:**
    > You can reset the Timeout by clicking on the **Reset Timeout**。
    >
    > 
    >

    d. Check SSH address, SSH port, console account, and console password.

    > **Tip:**
    > Click to view the password.



2. Enter access information on Putty.
    a. In the **Connection type** field, choose **SSH**.
    b. In the **Host Name (or IP address)** field, enter the SSH address of PBX.
    c. In the **Port** field, enter the SSH port of PBX.
    d. **Optional:** On the left navigation bar, click **Window > Lines of scrollback**, set a scrollback line number, so that you can get sufficient lines of log for debug analysis.
    e. Click **Open**.

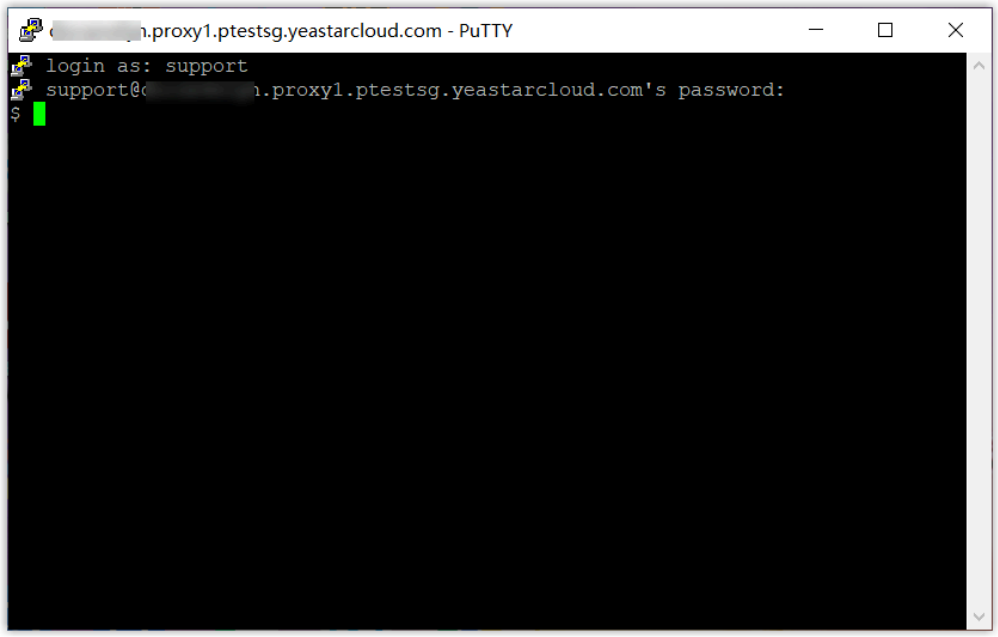


3. Verify your account and password.
    a. In the **login as** field, enter *support*.

b. Copy console password from PBX.
c. In the password field, right click to paste the password.

Result

If the following figure shows, you can successfully access and debug the PBX.



Allowed Country IPs

Restrict Specific Countries or Regions from Accessing Yeastar P-Series Cloud Edition

By default, all the countries and regions are allowed to access Yeastar P-Series Cloud Edition. Sometimes hackers may remotely access your phone system to make international and long-distance calls, monitor conversations, or do other operations that may cause security threats to your phone system. In this case, you can restrict specific countries or regions from accessing your phone system.

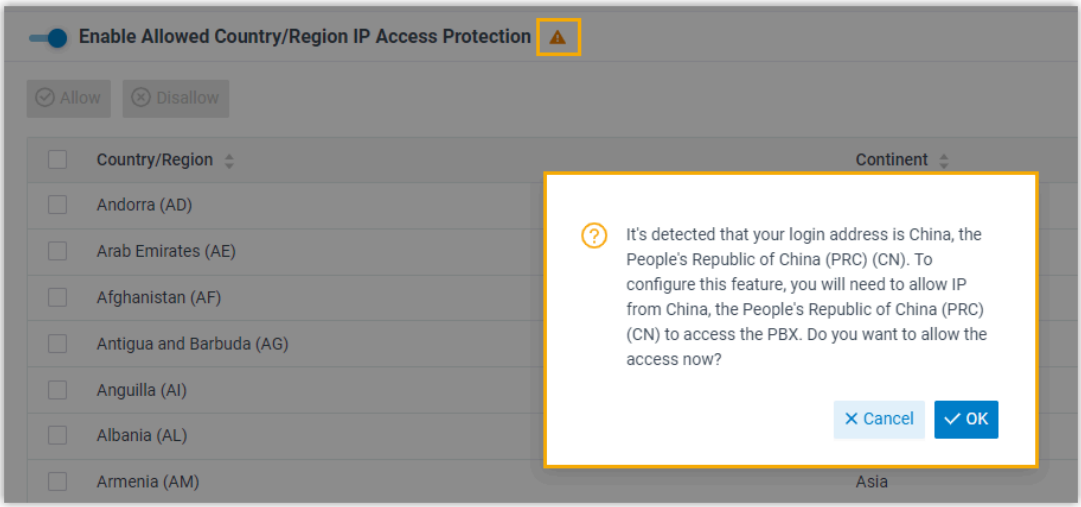
Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Allowed Country IPs**.

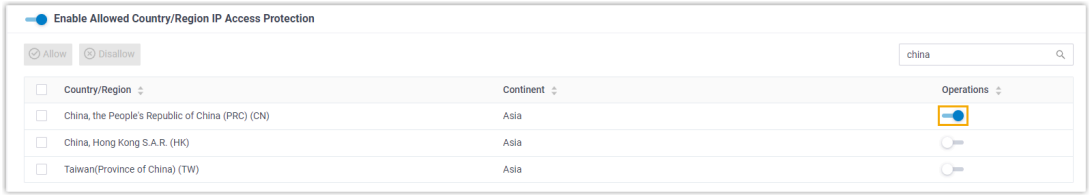
2. Turn on the option **Enable Allowed Country/Region IP Access Protection**.

   The system tries to identify whether the connection from your current IP address is accepted in the Allow IPs of your PBX.
3. If current IP address is not accepted in the Allow IPs of your PBX, the system would try to identify the country/region from which your current IP address is originated. You need to allow connections from your current country/region. Otherwise, you can NOT enable and use the IP access protection feature.

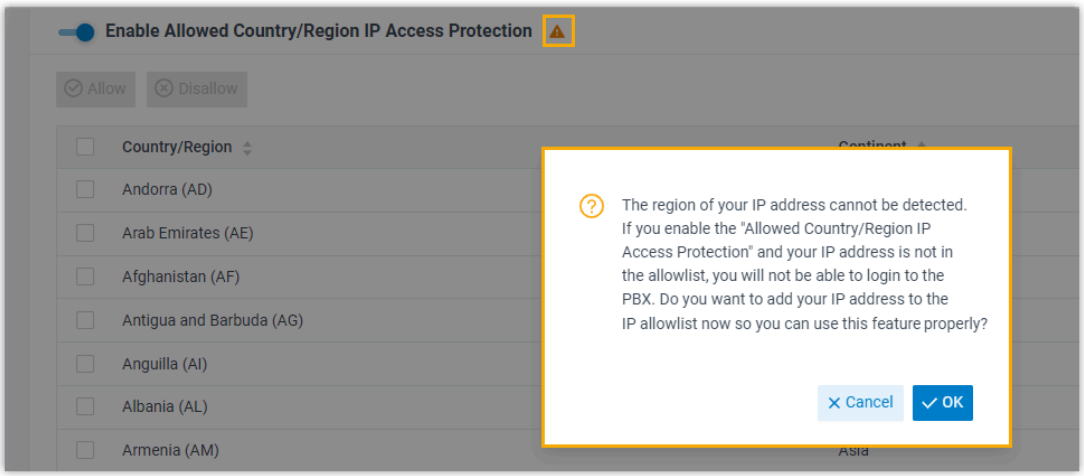
   - If the system successfully detects your country/region, a warning icon and a pop-up window will be displayed, prompting you to allow access for your current country/region.



   You should click **OK** to allow IPs from your current country/region to access the PBX. In doing so, the warning icon is disappeared, and operation status of your country/region is changed to .

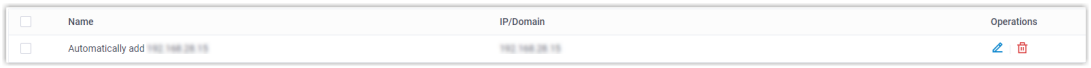


   - If the system fails to detect your country/region, a warning icon and a pop-up window will be displayed, prompting you to allow access for your current IP address.

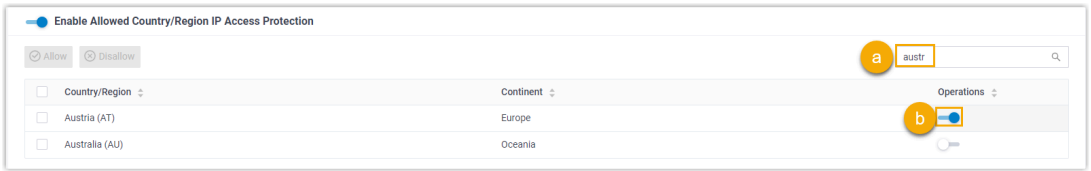


You should click **OK** to add your IP address to the Allow IPs. In doing so, the warning icon is disappeared, and connections from your IP address are accepted.

> **Tip:**
> You can check the added IP address on **Security > Security Rules > Allowed IPs**.



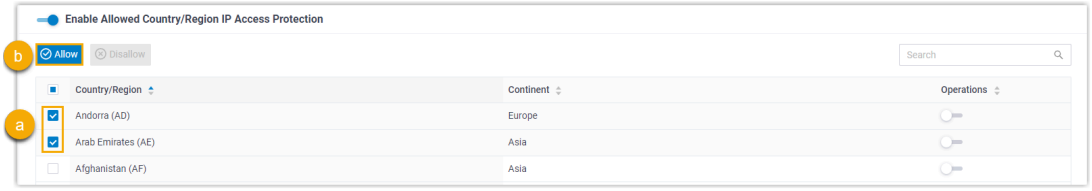
4. To allow desired countries/regions to access the PBX, do as follows:
   - To allow a country/region to access the PBX, do as follows:



     a. In the search box, enter a desired country or region.

     b. In the **Operations** column, set the status to .

   - To allow multiple countries/regions to access the PBX, do as follows:



a. Select the checkboxes of desired countries or regions.
b. Click **Allow**.

The status will be changed to .

5. Click **Apply**.

Result

Only the devices with IP addresses originating from the allowed countries or regions can access the PBX.

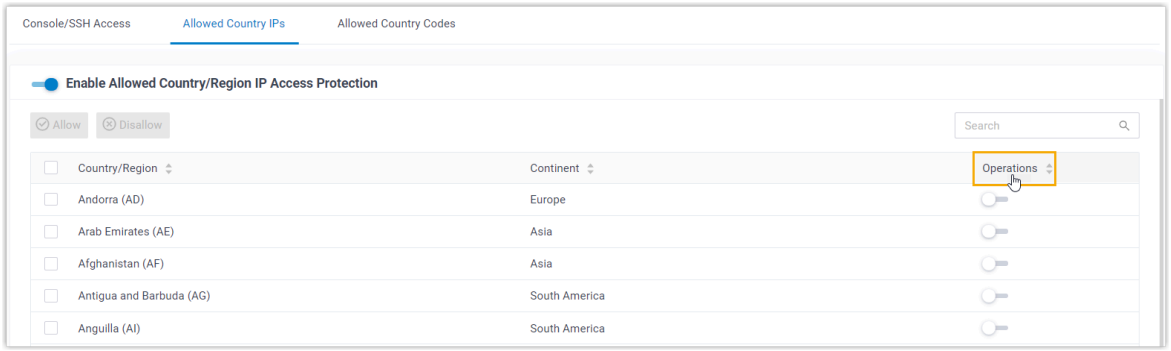
> **Tip:**
> For the disallowed countries or regions, if you want to allow a specific IP address to access the PBX, you can add the IP address to Allowed IPs list. For more information, see Add an Allowed IP Address.

Check Allowed Country/Region IP

By default, all the countries and regions are displayed in ascending (A to Z) alphabetical order, whether they are allowed to access Yeastar P-Series Cloud Edition or not. To check the allowed country/region IP, you need to sort all the countries and regions again.

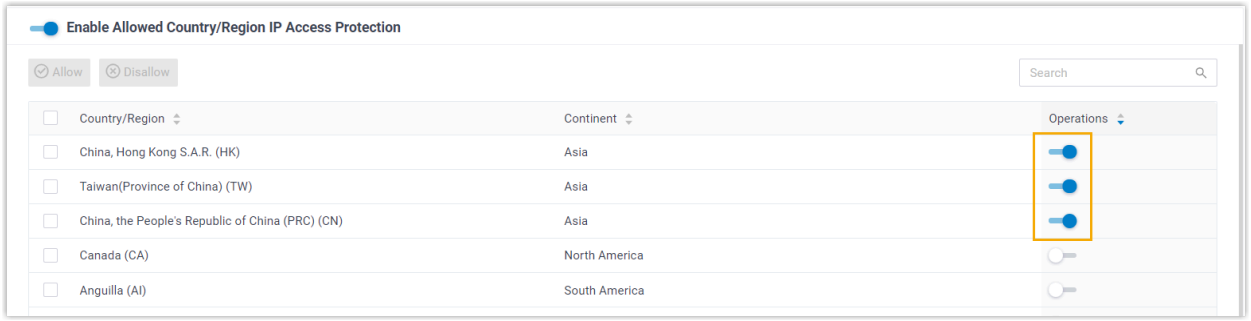
Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Allowed Country IPs**.
2. Click beside **Operations**.



Result

All the countries and regions that are allowed to access the PBX are moved to the top.



Allowed Country Codes

Restrict International Calls to Specific Countries or Regions

If there is an outbound route on your PBX that allows outbound international calls, the authorized users can make international calls to all the countries and regions. To prevent toll fraud, you can restrict users from making international calls to specific countries or regions.

Scenario

A manufacturer has a factory in Mexico, and his or her target customers are in Argentina. The manufacturer wants to restrict employees from making international calls to countries and regions except Argentina (country code 54).

Procedure

Based on the above scenario, you need to follow the instructions below to realize restrictions on international dialing:

- Step1. Allow international calls to Argentina only
- Step2. Allow employees to make international calls

Step1. Allow international calls to Argentina only

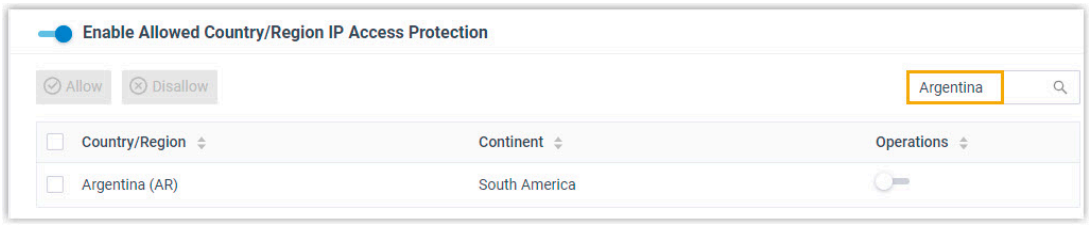
1. Log in to PBX web portal, go to **Security > Security Settings > Allowed Country Codes**.
2. Enable international dialing protection, and set international dialing code.

a. Turn on the option **Enable Allowed Country/Region Code Dialing Protection**.
b. In the **International Dialing Code** field, enter the prefix of international call according to your country. In the scenario, enter *00*.
When an employee tries to call a number starting with 00, the PBX's outbound route will identify this call as an international call.

> **Note:**
> Make sure there is at least one outbound route that matches with the international dialing code to route international calls out.

c. Click and **Apply**.

3. Set the countries or regions to which employees can make international calls.
a. In the search box, enter a desired country or region. In the scenario, enter *Argentina*.



b. In the **Operations** column, set the status to .

> **Note:**
> Some countries or regions share the same code (e.g. the country code for Canada and America is 1). If you allow international dialing to a country or a region, employees can also make calls to the countries or regions that share the same code.

c. Click **Apply**.

Step2. Allow employees to make international calls

By default, after you enable country/region code dialing protection, all the users are not allowed to make international calls. To allow employees to make international calls, you need to grant permission to desired employees.

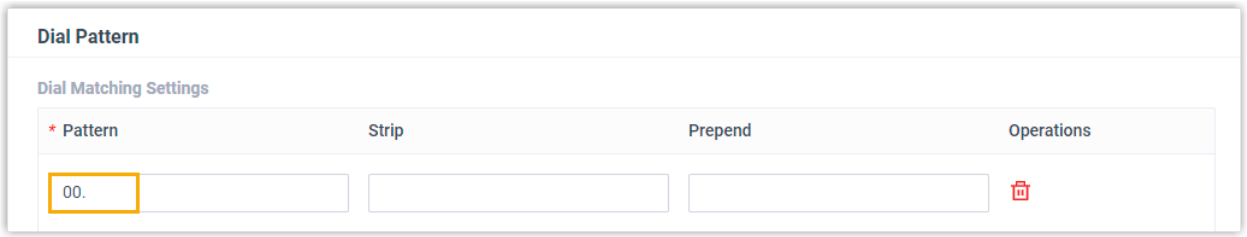
1. Go to **Extension and Trunk > Extension**.
2. Select the checkboxes of desired extensions, click **Edit**.

3. Click **Security** tab.
4. In the **Call Restrictions** section, select the checkbox of **Bulk Edit** and unselect the checkbox of **Disallow International Calls**.
5. Click **Save** and **Apply**.

Result

Authorized employees can make international calls to Argentina (country code 54).

The PBX has an outbound route configured as follows:



When an authorized employee dials a number, PBX's outbound route will check if the dialing is valid:

- When an authorized employee dials 00541938384, the dialing is considered as valid.
- When an authorized employee dials 00621938384, the dialing is considered as invalid.
- When an authorized employee dials 541938384, it will not be considered as an international dialing, and the PBX will check if there is a matched outbound route to route the call out.

Block Outbound International Calls

To restrict users from making international calls, you can restrict dial pattern of outbound routes, or set up international dialing protection. This topic describes how to set up international dialing protection to block outbound international calls.

Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Allowed Country Codes**.
2. Turn on the option **Enable Allowed Country/Region Code Dialing Protection**.
3. In the **International Dialing Code** field, enter the prefix of international call according to your country.

4. Click and **Apply**.

Result

All the extension users can NOT make international calls.

Passwordless Login

Allow Passwordless Login to Yeastar P-Series Cloud Edition

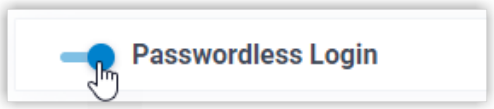
In case that you need the service provider to access your PBX system for remote technical support, and don't want to expose the administrator account credentials, you can set up the PBX to allow passwordless login. In this way, the service provider can directly log in to your PBX system without password.

Requirements

The firmware version of Yeastar P-Series Cloud Edition is 84.8.0.49 or later.

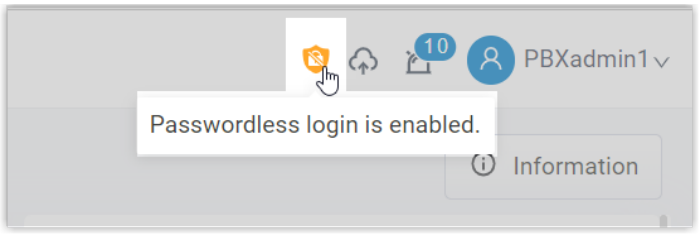
Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Passwordless Login**.
2. Turn on the **Passwordless Login** switch.

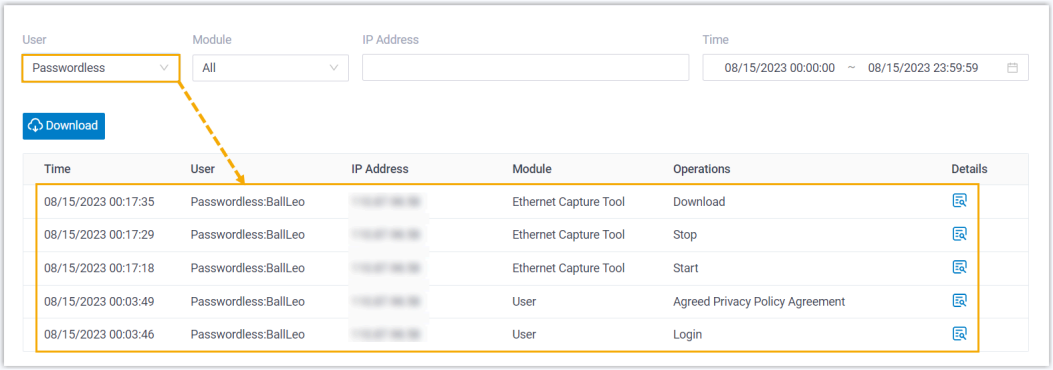


Result

- appears on the top-right corner of the PBX administrator portal (the web portal of super administrator), indicating that the 'Passwordless Login' feature is enabled.



- Service provider can directly log in to the PBX web portal with a dedicated **Password-less** account, which has the same permissions as the PBX super administrator.

> **Note:**
> The PBX system automatically records the operations of the **Passwordless** account, you can check the records on **Maintenance > Operation Logs**.
>
> 
>

Related information

[Disallow Passwordless Login to Yeastar P-Series Cloud Edition]

Disallow Passwordless Login to Yeastar P-Series Cloud Edition

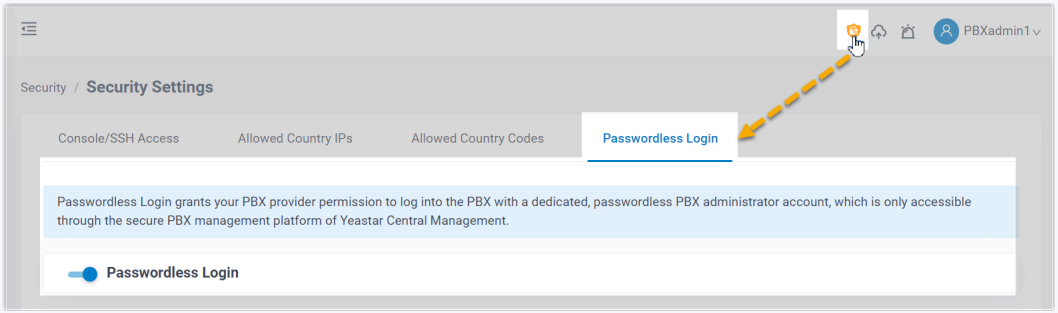
This topic describes how to disallow passwordless login to Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Passwordless Login**.

> **Tip:**

The super administrator can directly click on the top-right corner of PBX administrator portal to quickly access the **Passwordless Login** configuration page.



2. Turn off the **Passwordless Login** switch.

Result

on the top-right corner of the PBX administrator portal disappears; The service provider can NOT log in to the PBX system via passwordless login method.

Two-Factor Authentication (2FA)

Two-factor Authentication (2FA) Overview

Yeastar P-Series Cloud Edition supports to configure two-factor authentication (2FA) for super administrator account. With 2FA enabled, both the account password and an additional authentication code are required for login, which adds an extra layer of security to the PBX. This topic provides an overview of the supported two-factor authentication methods.

> **Note:**
> Extension users can also configure two-factor authentication for their own accounts on Linkus **Web Client** or **Desktop Client**, and the configuration will be applied to all their Linkus clients.
>
> For more information, see Two-factor Authentication (2FA) on Linkus Web Client and Two-factor Authentication (2FA) on Linkus Desktop Client.

Requirements

Yeastar P-Series Cloud Edition is 84.10.0.30 or later.

Two-factor authentication by authenticator application

This method requires you to install an authenticator application on your mobile phone. The supported applications are listed below:

- Google Authenticator
- FreeOTP
- Twilio Authy
- Microsoft Authenticator

After installing an authenticator application, you need to add your account to the application, via which you can obtain authentication codes for two-factor authentication. When you log in to the super administrator account, both account password and the authentication code generated by the authenticator application are required.

For more information about the configuration, see Configure Two-factor Authentication using Authenticator Application.

Two-factor authentication by Email

This method allows you to receive authentication codes for two-factor authentication via the email associated with the super administrator account. When you log in to the super administrator account, both account password and the authentication code sent to your email are required.

For more information about the configuration, see Configure Two-factor Authentication using Email.

Configure Two-factor Authentication using Authenticator Application

This topic describes how to configure two-factor authentication for super administrator account using an authenticator application on your mobile phone.

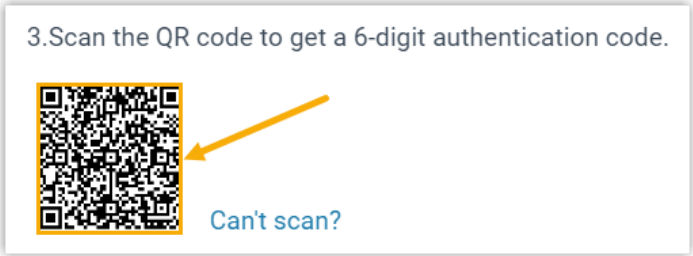
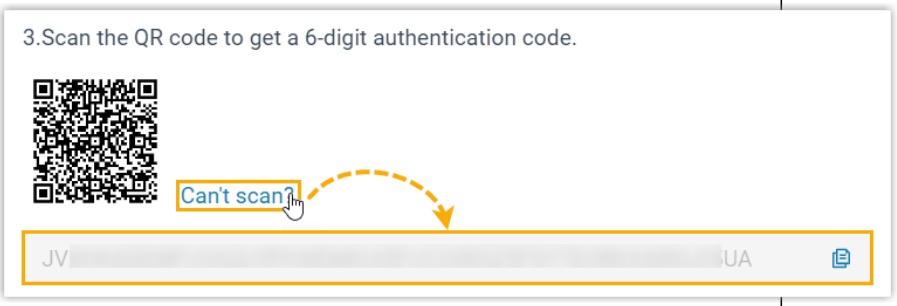
Prerequisites

You have installed one of the following supported authenticator applications on your mobile phone.

- Google Authenticator
- FreeOTP
- Twilio Authy
- Microsoft Authenticator

Procedure

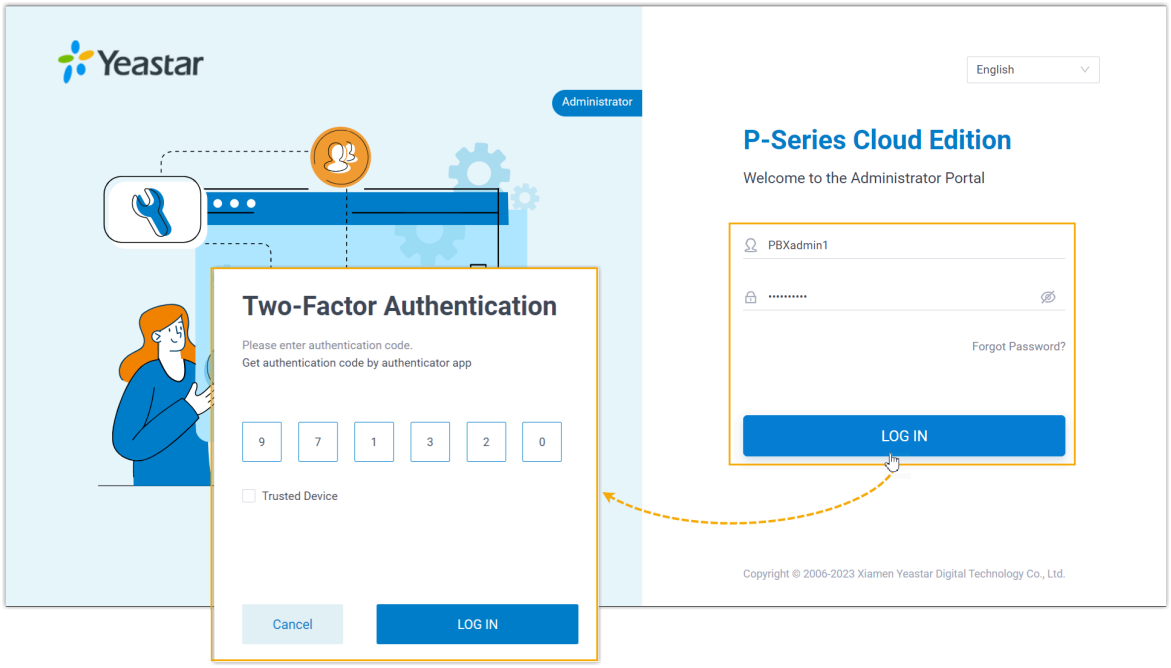
1. Log in to PBX web portal, click your account at the top-right corner, then go to **Change Password & Security > Security Settings**.
2. Select the checkbox of **Two-Factor Authentication**.
3. In the pop-up **Password** window, enter your account password and click **Confirm** to verify your operation.
4. Select **Authenticated by Authenticator**.
5. Add your account to the authenticator application via either of the following methods.

Method	Instruction
Scan QR Code to quickly add the account	You can quickly add your account to the authenticator application by scanning the QR code provided by PBX. a. On your mobile phone, open the authenticator application, and select to scan QR code. b. Scan the QR code shown on PBX web portal.  Your account is added to the application automatically, a 6-digit authentication code is shown.
Manually add the account	In case you can not scan QR code, you can manually add your account and enter the secret key provided by PBX. a. On PBX web portal, click **Can't scan** beside the QR code. PBX will generate a secret key, note it down for later use.  b. On your mobile phone, open the authenticator application, and select to manually add an account. c. Enter the relevant information, and paste the secret key. **Note:** If you need to complete more configurations for the secret key, you should set **SHA1** as the algorithm for **TOTP** protocol, and set to generate **6-digit** code with an interval of **30** seconds. Your account is added to the application, a 6-digit authentication code is shown.

6. On PBX web portal, enter the 6-digit authentication code in the **Authentication Code** field.
7. Click **Save**.

Result

- The webpage prompts a message "Edited successfully.", which means that you have successfully set up two-factor authentication.
- Next time you log in to PBX web portal, you need to enter an authentication code additionally.



Troubleshooting:
What if my extension users fail to log in with two-factor authentication?

You can disable two-factor authentication for their extension accounts, so that they can directly log in with their username and password.

Related information

Manage Two-factor Authentication of Super Administrator Account

Configure Two-factor Authentication using Email

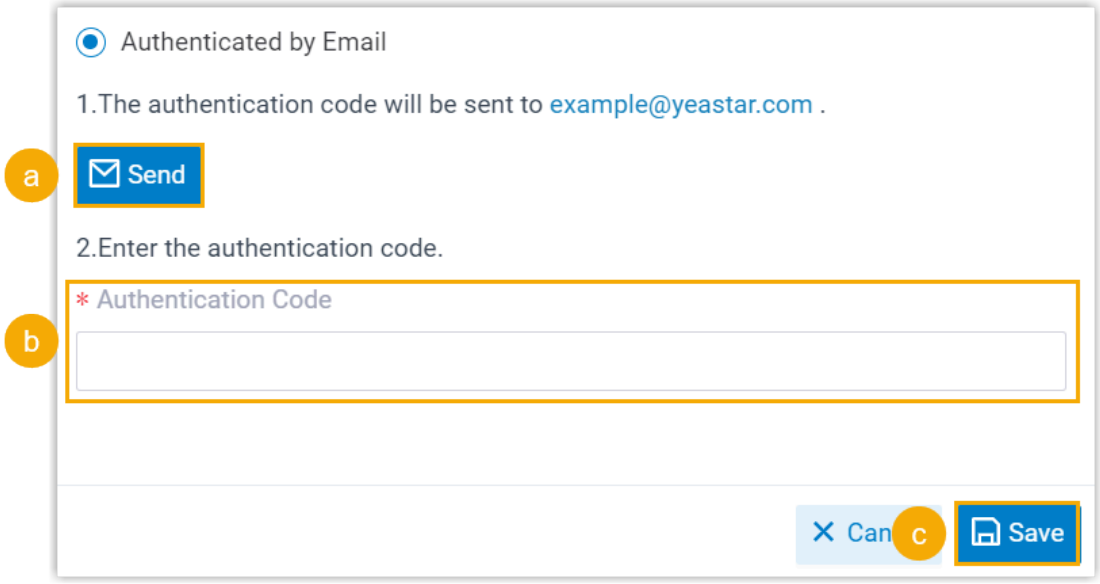
This topic describes how to configure two-factor authentication for super administrator account using email.

Prerequisites

System email server is set up.

Procedure

1. Log in to PBX web portal, click your account at the top-right corner, then go to **Change Password & Security > Security Settings**.
2. Select the checkbox of **Two-Factor Authentication**.
3. In the pop-up **Password** window, enter your account password and click **Confirm** to verify your operation.
4. Select **Authenticated by Email**, and complete the following settings:



   a. Click **Send**.
   An email containing a 6-digit authentication code is sent to the email address that is associated with the super administrator account.

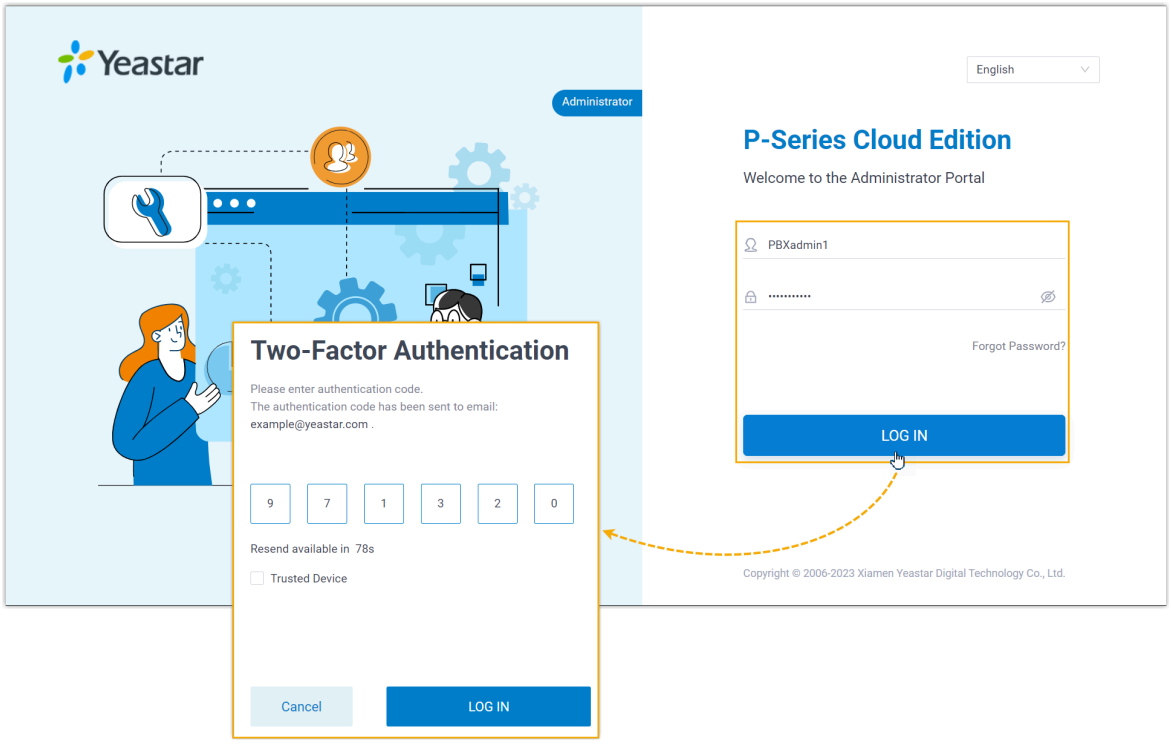
   > **Note:**
   > The code expires 5 minutes after the email is sent.

   b. In the **Authentication Code** field, enter the authentication code.

c. Click **Save**.

Result

- The webpage prompts a message "Edited successfully.", which means that you have successfully set up two-factor authentication.
- Next time you log in to PBX web portal, you need to enter an authentication code additionally.



Troubleshooting:
What if my extension users fail to log in with two-factor authentication?

You can disable two-factor authentication for their extension accounts, so that they can directly log in with their username and password.

Related information

Manage Two-factor Authentication of Super Administrator Account

Manage Two-factor Authentication of Super Administrator Account

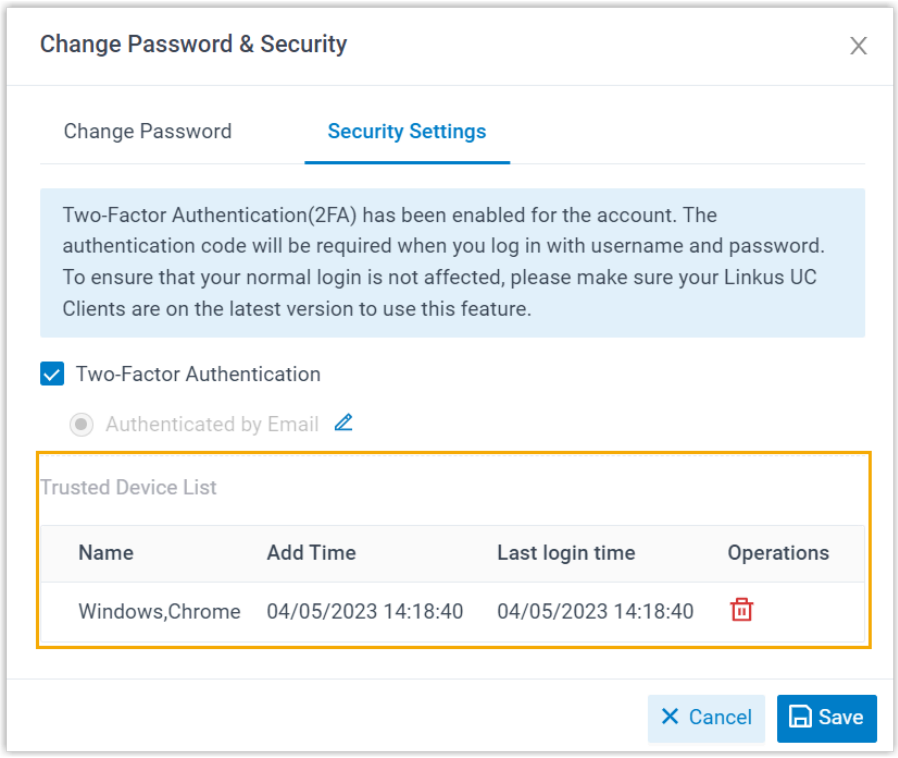
This topic describes how to manage the two-factor authentication feature for super administrator account, including removing trusted devices, changing authentication method, and disabling the two-factor authentication feature.

Remove a trusted device

As a super administrator, in case you lost access to a trusted device, you can remove it from the list if necessary.

1. Log in to PBX administrator portal with super administrator account, click your account at the top-right corner, then go to **Change Password & Security > Security Settings**.

   The trusted devices are displayed in the **Trusted Device List** section.



2. Click 🗑 beside the device that you want to remove.
3. In the pop-up window, click **OK**.

Change two-factor authentication method

You can change the two-factor authentication method for your super administrator account as needed.

1. Log in to PBX administrator portal with super administrator account, click your account at the top-right corner, then go to **Change Password & Security > Security Settings**.
2. Click beside the current authentication method.
3. Select the desired method, then complete the follow-up settings accordingly.

Disable two-factor authentication

Disable two-factor authentication for your account

You can disable two-factor authentication for your super administrator account at any time.

1. Log in to PBX administrator portal with super administrator account, click your account at the top-right corner, then go to **Change Password & Security > Security Settings**.
2. Unselect the checkbox of **Two-Factor Authentication**.
3. In the pop-up **Password** window, enter your account password and click **Confirm** to verify your operation.
4. In the **Security Settings** tab, click **Save**.

   The webpage prompts a message "Edited successfully.", which means that you have successfully disabled two-factor authentication.

Disable two-factor authentication for your extension user

If your extension user lost access to their two-factor authentication (e.g. they lost their authenticator device or could not receive authentication code via email), you can disable the two-factor authentication for their extension accounts, so that they can directly log in with username and password.

1. Log in to PBX web portal, go to **Extension and Trunk > Extension**.
2. Click beside the extension, then click the **Security** tab.
3. Scroll down to the **Login Security** section, then unselect the checkbox of **Two-Factor Authentication**.

4. In the pop-up **Password** window, enter your account password and click **Confirm** to verify your operation.
5. Click **Save**.

   The two-factor authentication of the extension account is disabled.

Related information

Configure Two-factor Authentication using Authenticator Application
Configure Two-factor Authentication using Email
Enforce Two-factor Authentication for All Extension Users

Enforce Two-factor Authentication for All Extension Users

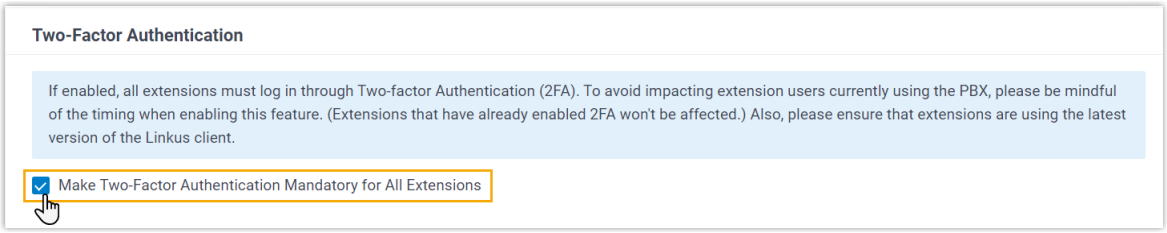
To enhance security for users' accounts, you can make Two-Factor Authentication (2FA) mandatory for all extension users. In this way, users will be required to go through two-factor authentication when logging in using their extension username and password, thus preventing unauthorized access.

Requirements

- Yeastar P-Series Cloud Edition is 84.14.0.24 or later.
- Linkus Desktop Client is 1.4.9 or later.

Procedure

1. Log in to PBX web portal, go to **Security > Security Settings > Security Options**.
2. In the **Two-Factor Authentication** section, select the checkbox of **Make Two-Factor Authentication Mandatory for All Extensions**.



3. Click **Save**.

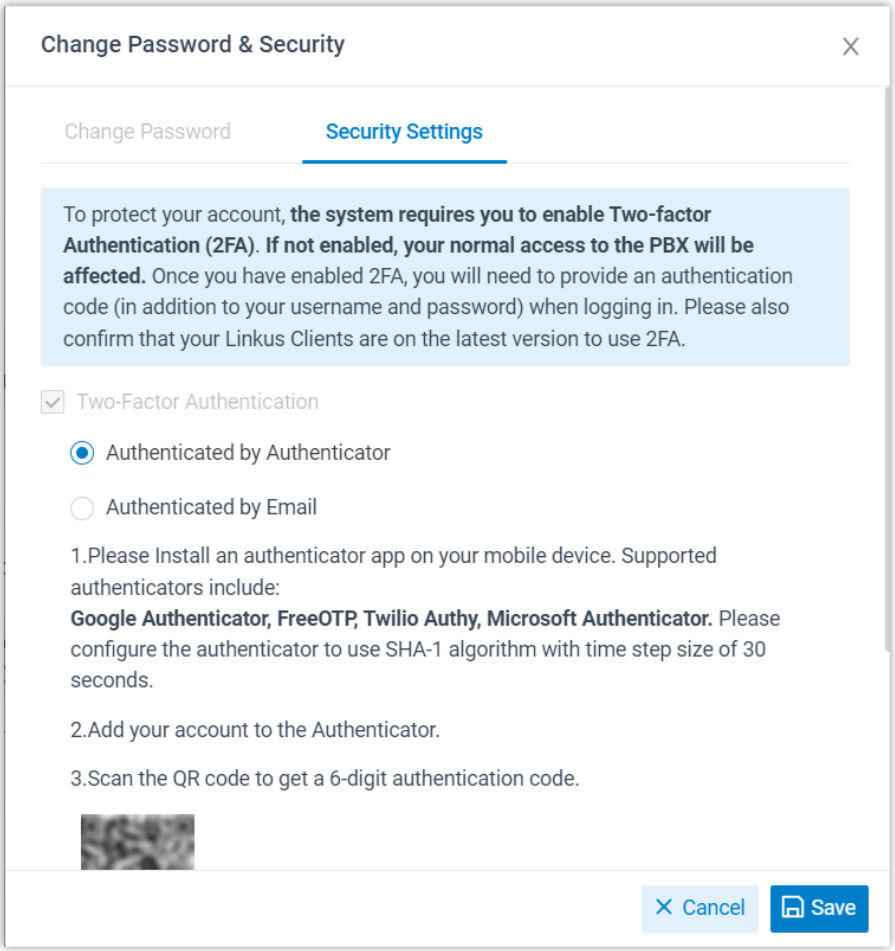
Result

The result varies depending on whether the two-factor authentication (2FA) feature is enabled on user's Linkus clients:

- **2FA is already enabled on users' Linkus clients**

  These users will not experience any interruption in using Linkus.
- **2FA is disabled on users' Linkus clients**

  Users who attempt to log in or are already logged in Linkus Web Client or Desktop Client using their extension username and password, the two-factor authentication setup window will pop up, requiring the users to complete the 2FA setup for their accounts. Otherwise, their accounts will be automatically logged out.



Maintenance

Upgrade

Check for Available Firmware Updates

This topic describes how to automatically or manually check for firmware updates.

> **Note:**
> If you couldn't find the **Upgrade** feature, contact your PBX provider if needed.

Automatic check for firmware updates

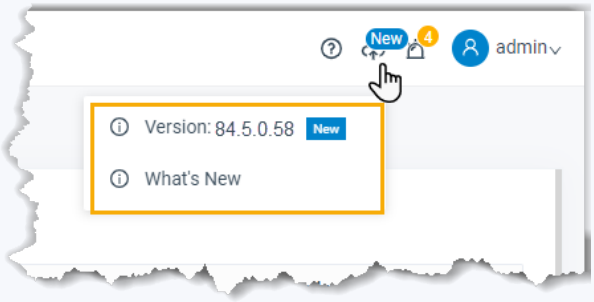
Procedure

1. Log in to PBX web portal, go to **Maintenance > Upgrade**.
2. In the **Automatic Upgrade** section, select **Check for updates and notify me**.
3. In the **Automatically check for updates at** drop-down list, set when the system should check for new version. This can be a daily or weekly check.
4. Click **Save**.

Result

The system will regularly check for new firmware.

> **Note:**
> If a new firmware is detected, the followings can be achieved:
>
> - At the top-right corner of PBX web portal, 
>
> is displayed. Click 
>
> , click **What's New** to check release notes for the new version or click **Version** to go to upgrade page.



- The system will notify you via specific notification methods if you have enabled notification for **New System Firmware Detected** event.

Manual check for firmware updates

Procedure

1. Log in to PBX web portal, go to **Maintenance > Upgrade**.
2. Click **Check for the New Firmware**.

Result

If a new firmware is detected, you will find a table as below. Click the link under **Release Notes** to check what's updated in the new version, and decide whether to upgrade the firmware now.

Version	Release Notes	Upgrade
37.1.0.13	https://help.yeastar.com/en/p-series-appliance-edition/release-notes/v37.1.0.13.html	Upgrade Now

Version	Release Notes	Upgrade
84.5.0.58	https://help.yeastar.com/en/p-series-cloud-edition/release-notes/v84.5.0.58.html	Upgrade now

Backup and Restore

Overview of Backup and Restore

Yeastar P-Series Cloud Edition supports to back up configuration data, and restore data on the same PBX or another PBX.

How Backup and Restore feature benefits your work

Yeastar P-Series Cloud Edition integrates backup and restore feature, which helps you achieve the followings:

- Create regular and scheduled backups.
- Easy data transfer from one PBX to another.
- Quick restoration and recovery in case of system failure.

Backup data

Yeastar P-Series Cloud Edition supports to back up the following configuration data:

- **System Configuration**: All the configurations on the system.
- **Custom Prompts**
- **CDR**
- **Company Contacts and Phonebooks Settings**
- **Chat Data for External Chat**

Backup locations

Backup files are stored in a default storage area, and can NOT be changed.

Backup file cleanup

By default, when the number of backup files reaches 5, the oldest files will be replaced by the newest files. You can retain the default value, or change the value according to your needs.

For more information, see Auto Cleanup Settings.

Backup and restore logs

The PBX always makes records whoever backs up or restores the PBX configuration data, you can check the operation details on PBX web interface.

For more information, see Manage Operation Logs.

Create an On-Demand Backup

This topic describes how to manually back up PBX configurations.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Backup and Restore**, click **Backup**.
2. Configure backup settings.
    - **File Name**: Retain the default name or enter a name to help you identify it.
    - **Comments**: Add a note to the backup file.
    - **The backup file will include**: Select the items that will be backed up.
        - **System Configuration**: All the configurations on the system.
        - **Custom Prompts**
        - **CDR**
        - **Company Contacts and Phonebooks Settings**
        - **Chat Data for External Chat**
3. Click **Save**.

Result

The created backup file is displayed in **Backup and Restore** list and is stored in the selected location.

> **Tip:**
> You can archive the backup file to external servers via FTP, SFTP, Amazon S3, and Google Cloud Storage services for less space occupying and minimized data loss risk on PBX, as well as easier file management on external server. For more information, see Yeastar P-Series Cloud Edition Remote Archiving Overview.

Set up an Automatic Backup Schedule

Yeastar P-Series Cloud Edition supports to automatically back up specific configuration data at the scheduled time. This topic describes how to set up an automatic backup schedule.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Backup and Restore**, click **Backup Schedule**.
2. In the pop-up window, enable **Backup Schedule**.
3. Configure an automatic backup schedule.
    a. Set the automatic backup period. This can be a daily, weekly, or monthly backup.

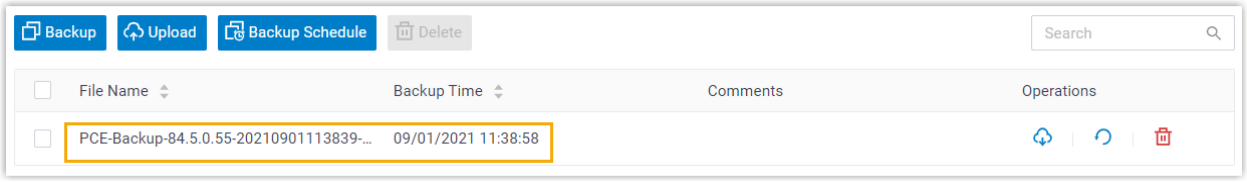
- **Frequency**: Choose to make a daily, weekly, or monthly backup.
  - **Daily**: If you choose the option, select a time from the drop-down list. The system backs up the settings at this time of the day.
  - **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system backs up the settings at this time of the week.
  - **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system backs up the settings on this day and time of the month.

b. In the **The backup file will include** section, choose the items that will be backed up.
   - **System Configuration**: All the configurations on the system.
   - **Custom Prompts**
   - **CDR**
   - **Company Contacts and Phonebooks Settings**
   - **Chat Data for External Chat**

4. Click **Save**.

Result

The system will back up the specified configuration data at the scheduled time. The automatic generated backup file will be displayed in the **Backup and Restore** list.



Tip:
You can archive the backup file to external servers via FTP, SFTP, Amazon S3, and Google Cloud Storage services for less space occupying and minimized data loss risk on PBX, as well as easier file management on external server. For more information, see Yeastar P-Series Cloud Edition Remote Archiving Overview.

Restore Your System from a Backup

In case of data loss or system failure, you can restore the PBX from a backup. This topic describes how to restore data on the PBX.

Prerequisites

- Make sure that you have backed up system configurations and required files, such as custom prompts and CDR.
- Read and understand restrictions for data restoration.
  - You can restore a backup file that is created from an older version of PBX.

    Example: Restoring a backup file (v84.5.0.55) to PBX (v84.5.0.57) would work.
  - You can NOT restore a backup file that is created from a newer version of PBX.

    Example: Restoring a backup file (v84.5.0.57) to PBX (v84.5.0.55) would NOT work.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Backup and Restore**.
2. Select a backup file to which you want to restore, click  .
3. In the pop-up dialog box, click **OK**.
4. Reboot the PBX to take effect.

Result

The current configurations on your PBX are OVERWRITTEN with the backup data.

Related information

Create an On-Demand Backup
Set up an Automatic Backup Schedule

Reboot

Reboot Yeastar P-Series Cloud Edition on Web Interface

This topic describes how to reboot Yeastar P-Series Cloud Edition on web interface.

Prerequisites

Make sure there aren't ongoing calls, or the calls would be disconnected.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Reboot**.
2. In the **Reboot Now** section, click **Reboot Now**.
3. In the pop-up dialog box, click **Yes** to reboot the PBX.

Result

It takes about one minute to reboot the system.

If you have enabled notification for **System Reboot** event, the system will inform relevant contacts of the reboot via specific notification methods.

Reset

Reset the System on Web Interface

This topic describes how to reset Yeastar P-Series Cloud Edition on web interface. After you reset the system, the domain name of Yeastar P-Series Cloud Edition does not change, but you need to re-activate the PBX.

Prerequisites

- Make sure there aren't ongoing calls, or the calls would be disconnected.
- We recommend that you create a backup file for PBX configurations.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Reset**.
2. Set which configurations and data that you want to clear.
    - **Reset All**: Clear all the configurations and data on the PBX.
    - **Reset CDR**: Clear all call logs.
    - **Reset Backup Files**: Clear backup files.
    - **Reset Prompts**: Clear custom prompts.

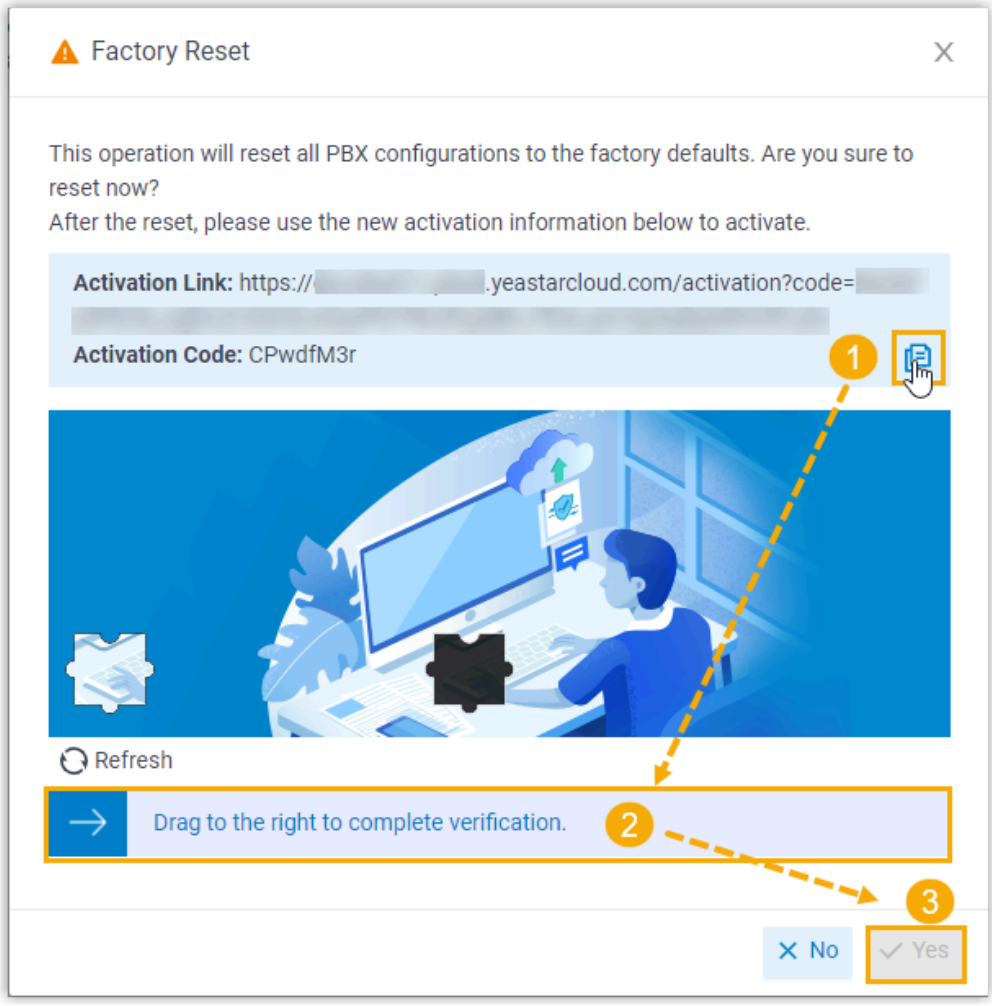
> **Note:**
> Whether the option is enabled or not, system prompts, music on hod, and preference settings for all the prompts would be cleared.

- **Reset Company Contacts**: Clear company contacts, phonebooks, and Caller ID match settings.
- **Reset Other System Configurations**: Reset all the logs and configurations except CDR, backup files, prompts, and contacts.

3. Click **Factory Reset**.

   A **Factory Reset** dialog box pops up.

4. In the pop-up dialog box, complete the followings.



   a. Click to take note of the **Activation Link** and **Activation Code**.

> **Important:**
> You need to use the new activation link and activation code to activate the PBX after resetting.

b. Drag to verify your operation.
c. Click **Yes**.

Result

It takes several minutes to reset the PBX. After resetting, the current web page prompts that the PBX is not activated.

What to do next

- Click the activation link, you are redirected to the PBX activation page. Paste the activation code and click **Confirm**.
- You are redirected to Installation Wizard page, follow the Installation Wizard to set up the PBX.

Operation Logs

Operation Logs Overview

This topic describes what are operation logs, what operations are recorded, and introduces the storage and auto cleanup of operation logs.

What are operation logs

Operation logs record successful operations performed on Yeastar P-Series Cloud Edition, and provide you with the followings to help you monitor and analyze the causes of systems errors or other types of problems.

- **Who**: Check who performed the operation. You can query all users' operations, or query operations by administrator or a specific extension.
- **When**: Check when the operation was performed. You can query operations by specific date and time.
- **What**: Check what operation was performed.

- **Where**: Check on which module the operation was performed. You can query operations by a specific module.

Storage of operation logs

Operation logs are saved in a default storage area, you can NOT change the storage location.

Auto cleanup of operation logs

By default, when operation logs reach 50,000, the system automatically deletes the oldest logs. You can change the value, or set the maximum days that logs can be retained. For more information, see Auto Cleanup Settings.

Note:
A few logs related with system security and user privacy are RETAINED so that Yeastar Support can help you troubleshoot problems when toll fraud happens or PBX suffers from attack.

The operation logs that will NOT be automatically cleaned up are as follows:

Table 48.

Event Type	Event
Operation	Administrator Login Success
	Administrator Password Changed
	Web User Login Success
	Web User Login Failed
	Linkus Client Login Failed
	Extension User Password Changed
Telephony	Emergency Call Dialed Out
System	Yeastar SMTP Server Error
Security	Web User Locked Out
	Linkus User Blocked Out
	Extension Registration Blocked Out
	Auto Defense IP Blocked Out
	Outbound Call Frequency Exceeded

Table 48. (continued)

Event Type	Event
	Outbound Call to a Disallowed Country

Manage Operation Logs

This topic describes how to view and download operation logs on Yeastar P-Series Cloud Edition.

View operation logs

1. Log in to PBX web portal, go to **Maintenance > Operation Logs**.
2. Set the filter criteria.
   - **User**: Query all users' operations, or query operations by administrator or a specific extension.
   - **Module**: Query operations on all modules, or query operations by a specific module.
   - **IP Address**: Query operations by the originated IP address.
   - **Time**: Query operations by specific date and time.
3. **Optional:** Click beside the desired log to check operation details.

Download operation logs

1. Log in to PBX web portal, go to **Maintenance > Operation Logs**.
2. To download all the operation logs, click **Download**.
3. To download the filtered operation logs, set the filter criteria, click **Download**.

   Logs are exported to a CSV file.

Troubleshooting

Capture Network Packet

This topic describes how to capture network packets of your PBX.

Background information

Ethernet Capture Tool may be required to capture packets in the following situations:

- Extension registration failure.
- No audio or one-way audio during a call.
- Occasional VoIP interconnection failure.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Troubleshooting > Ethernet Capture Tool**.
2. Click **Start**.

   The PBX starts to capture the Ethernet packet. During the time period, you should reproduce the problem of your VoIP trunks or extensions.
3. Click **Stop** to stop capturing.

   The packets are intercepted and saved on PBX's local flash.
4. Click **Download** to download the captured packet.

What to do next

Decompress the `.tar` file and use **Wireshark** software to open the packet file.

Use IP Ping Tool to Diagnose Network Issues

This topic describes how to use IP Ping tool to test if Yeastar P-Series Cloud Edition can reach a specific hostname or IP address, and introduces the test result.

Background information

Based on the Internet Control Message Protocol (ICMP), IP Ping is a network tool to determine if a destination server is accessible and estimate how long a packet takes to send and receive data from the server.

If you are suffering from the followings, you can use IP Ping to diagnose:

- Network issues.

  For example, you can use IP Ping to check if the PBX can access to a specific IP address or domain name.
- Poor VoIP call quality.

For example, if you are experiencing echo, buzzing, or latency during a call, you can use IP Ping to check jitter and latency, or if there are any packet loss.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Troubleshooting > IP Ping**.
2. In the **Target Host** field, enter the target domain or IP address.
3. Click **Start**.
4. Click **Stop** as your need.

Read the output

Example1: A successful Ping

```
start...
PING 192.168.6.11 (192.168.6.11): 56 data bytes
64 bytes from 192.168.6.11: seq=0 ttl=64 time=8.853 ms
64 bytes from 192.168.6.11: seq=1 ttl=64 time=0.778 ms
64 bytes from 192.168.6.11: seq=2 ttl=64 time=1.394 ms

--- 192.168.6.11 ping statistics ---
3 packets transmitted, 3 packets received, 0% packet loss
round-trip min/avg/max = 0.778/3.675/8.853 ms
```

The above example shows the followings:

- The device sends 3 ping packets and receives response for all the 3 packets.
- The ping packets' size are 64 bytes.
- The TTL value is 64, which indicates that packets are always forwarded to the same region.
- The time indicates that how long it takes to receive an Echo Response message after an Echo Request message is sent. This parameter can be used as a reference to determine whether the network is congested.

Example2: A failed Ping

```
start...
PING 192.168.7.2 (192.168.7.2): 56 data bytes

--- 192.168.7.2 ping statistics ---
```

```
60 packets transmitted, 0 packets received, 100% packet loss
```

The above example indicates that there is an issue of either the connection or the target device.

Use Traceroute Tool to Diagnose Network Issues

This topic describes how to use Traceroute tool to trace routes to a specific hostname or IP address, and introduces test results.

Background information

Traceroute is a network tool that tracks the gateways that packets pass through from Yeastar P-Series Cloud Edition to a destination server and helps you check network connectivity and locate network faults.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Troubleshooting > Traceroute**.
2. In the **Target Host** field, enter the target domain or IP address.
3. Click **Start**.

   The PBX starts to trace routes to the target domain or IP address.
4. Click **Stop**, or the traceroute will terminate automatically when completed.

Read the output

Example1: A good traceroute

```
start...
traceroute to www.baidu.com (36.152.44.95), 30 hops max, 46 byte
 packets
 1  *  *  *
 2  *  *  *
 3  192.168.1.1 (192.168.1.1)  1.853 ms  11.642 ms  19.951 ms
 4  110.80.36.161 (110.80.36.161)  3.008 ms  2.966 ms  3.943 ms
 5  61.154.238.133 (61.154.238.133)  7.369 ms  27.982 ms  7.808
 ms
 6  117.30.27.177 (117.30.27.177)  6.125 ms  117.30.24.213 (117.
30.24.213)  4.664 ms  4.376 ms
 7  202.97.36.117 (202.97.36.117)  26.446 ms  202.97.64.178 (202
.97.64.178)  22.534 ms  202.97.79.33 (202.97.79.33)  20.897 ms
```

```
 8  202.97.63.18 (202.97.63.18)  33.276 ms  202.97.76.238 (202.9
7.76.238)  36.685 ms  202.97.18.46 (202.97.18.46)  33.961 ms
 9  *  *  *
10  *  *  *
11  *  *  *
12  *  *  *
13  *  *  *
14  221.183.14.14 (221.183.14.14)  40.599 ms  221.183.18.2 (221.
183.18.2)  54.233 ms
15  21.22.207.183.static.js.chinamobile.com (183.207.22.21)  43.
056 ms  53.602 ms  50.481 ms
16  122.23.207.183.static.js.chinamobile.com (183.207.23.122)  4
7.251 ms  126.23.207.183.static.js.chinamobile.com (183.207.23.1
26)  47.401 ms  110.23.207.183.static.js.chinamobile.com (183.20
7.23.110)  54.380 ms
17  *  *  *
18  *  *  *
19  *  *  *
20  *  *  *
21  *  *  *
22  *  *  *
23  202.97.23.149 (202.97.23.149)  14.133 ms  *  202.97.23.157 (
202.97.23.157)  28.851 ms
24  61.154.238.69 (61.154.238.69)  7.096 ms  117.30.24.213 (117.
30.24.213)  4.682 ms  117.30.27.189 (117.30.27.189)  2.758 ms
25  113.96.4.170 (113.96.4.170)  14.663 ms  113.96.5.118 (113.96
.5.118)  17.857 ms  113.96.4.190 (113.96.4.190)  20.665 ms
26  *  *  *
27  *  *  *
28  *  *  *
29  110.80.36.161 (110.80.36.161)  4.278 ms  2.696 ms  3.900 ms
30  61.154.238.133 (61.154.238.133)  11.424 ms  4.690 ms  7.770
 ms
```

The above example displays in the format of `HOP Domain Name (IP Address) RTT1 RTT2 RTT3`.

- `Hop`: Whenever a packet is passed between a router, this is referred to as a “hop.” For example, in the output above, we can see that it takes 14 hops to reach **www.baidu.com** from the current location.
- `Domain Name [IP Address]`: The domain name, if available, often helps you see the location of a router. If this is unavailable, only the IP address of the router is displayed.

- `RTT1,RTT2,RTT3`: This is the round-trip time that it takes for a packet to get to a hop and back to your computer (in milliseconds). This is often referred to as latency, and is the same number you see when using ping. Traceroute sends three packets to each hop and displays each time, so you have some idea of how consistent (or inconsistent) the latency is. If you see a * in some columns, you didn't receive a response - which could indicate packet loss.

Example2: A failed hop

```
start...
traceroute to www.baidu.com (14.215.177.38), 30 hops max, 46 byt
e packets
 1  *  *  *
 2  *  *  *
 3  192.168.1.1 (192.168.1.1)  1.702 ms  4.912 ms  1.873 ms
 4  110.80.36.161 (110.80.36.161)  16.068 ms  2.642 ms  2.705 ms
 5  61.154.238.129 (61.154.238.129)  5.405 ms  61.154.238.133 (6
1.154.238.133)  9.038 ms  61.154.238.129 (61.154.238.129)  4.084
 ms
 6  117.30.27.185 (117.30.27.185)  3.183 ms  117.30.24.213 (117.
30.24.213)  5.256 ms  29.543 ms
 7  202.97.19.125 (202.97.19.125)  23.899 ms  202.97.23.153 (202
.97.23.153)  15.059 ms  202.97.21.69 (202.97.21.69)  12.542 ms
 8  113.96.4.130 (113.96.4.130)  20.978 ms  113.96.4.54 (113.96.
4.54)  17.600 ms  113.96.4.102 (113.96.4.102)  18.980 ms
 9  113.96.4.209 (113.96.4.209)  18.324 ms  25.160 ms  106.96.13
5.219.broad.fs.gd.dynamic.163data.com.cn (219.135.96.106)  29.13
5 ms
10  14.29.117.242 (14.29.117.242)  22.918 ms  121.14.67.150 (121
.14.67.150)  15.187 ms  14.215.32.126 (14.215.32.126)  15.963 ms
11  *  *  *
12  *  *  *
13  *  *  *
14  *  *  *
15  *  *  *
16  *  *  *
17  *  *  *
18  *  *  *
19  *  *  *
20  *  *  *
21  *  *  *
22  *  *  *
23  *  *  *
```

```
24  *  *  *
25  *  *  *
26  *  *  *
27  *  *  *
28  *  *  *
29  *  *  *
30  *  *  *
```

In the above example, hop1 and hop2 do not respond to the request, but they forward traffic to hop3. The test fails at hop11, and continues to fail all the way to hop30 (the max hops).

Enable Core Call Service Anomaly Detection

Anomaly in core call service will cause registration dropouts and call failures. Yeastar P-Series Cloud Edition can minimize the impact by automatically detecting the health of core call service and initiating a recovery process when any abnormalities are detected.

Requirements

The firmware version of Yeastar P-Series Cloud Edition is 84.15.0.22 or later.

Procedure

1. Log in to PBX web portal, go to **Maintenance > Troubleshooting > Advanced**.
2. Turn on the option **Core Call Service Anomaly Detection**, then configure the following settings:
   - **Detection Interval (s)**: Enter the interval between each detection to determine how frequently the system will check for core call service anomaly.

     > **Note:**
     > The value you enter should not be smaller than 10.

   - **Anomaly Confirmation Delay (s)**: Enter the waiting time before confirming a core call service anomaly after it is detected.

     > **Note:**
     > The value you enter should not be smaller than 30.

3. **Optional:** To allow the system to automatically initiate a recovery process after confirming a core call service anomaly, select the checkbox of **Automatic Recovery from Abnormality**.

   > **Note:**
   > During the recovery, all ongoing calls will be disconnected.

4. Click **Save** and **Apply**.

Result

The system will periodically check for the health of core call service.

- If core call service is found abnormal, the **Abnormal Core Call Services** event will be triggered to notify the relevant contacts.
- If automatic recovery is enabled, the system will automatically initiate a recovery process when any abnormalities are detected. Once the core call service is successfully recovered, the **Core Call Services Recovery Completed** event will be triggered to notify the relevant contacts.

> **Note:**
> If the core call service is not recovered, you can download system logs and contact Yeastar for support.

System Logs

System Logs Overview

This topic describes what are system logs, which level's logs are recorded, and introduces the storage and auto cleanup of system logs.

What are system logs

System logs are log files that contain information about system activities, which helps you troubleshoot and debug the system. The daily-generated system logs are displayed on **System Logs**, you can view and download logs on PBX web portal.

Log levels

Yeastar P-Series Cloud Edition provides multiple log levels, each of them records different information. The supported log levels are as follows:

- **Information**: Basic information.
- **Notice**: Notice information.
- **Warning**: Warning information.
- **Error**: Error information.
- **DTMF**: DTMF information.
- **Time Log**: Add time stamp of system logs.
- **Debug**: Debug information.
  - **Enable SIP Debug**
  - **Enable RTP Debug**

Storage of system logs

System logs are saved in a default storage area, you can NOT change the storage location.

Auto cleanup of system logs

By default, the system automatically deletes the oldest system logs every 7 days, or when logs reach 10MB. You can change the maximum file size or days that logs can be retained. For more information, see Auto Cleanup Settings.

Related information

Configure Log Level
Manage System Logs

Configure Log Level

Yeastar P-Series Cloud Edition allows you to configure log level to gather only information that you consider important. This topic describes how to configure log level.

Procedure

1. Log in to PBX web portal, go to **Maintenance > System Logs**.
2. Click **Log Level**.
3. In the pop-up window, decide which level's logs that you want to trace.
   - **Information**: Basic information.

- **Notice**: Notice information
- **Warning**: Warning information.
- **Error**: Error information.
- **DTMF**: DTMF information.
- **Time Log**: Add time stamp of system logs.
- **Debug**: Debug information.
  - **Enable SIP Debug**
  - **Enable RTP Debug**

4. Click **Save** and **Apply**.

Result

The system will generate logs of the specified levels every day.

Manage System Logs

This topic describes how to download or delete system logs on Yeastar P-Series Cloud Edition.

Download system logs

1. Log in to PBX web portal, go to **Maintenance > System Logs**.
2. Download one or more system logs according to your needs.
   - To download a system log, click  beside the desired log.
   - To bulk download system logs, select the checkboxes of the desired logs, click **Download**.

   The desired logs are downloaded and compressed into a `.tar` file.

> **Tip:**
> You can decompress the file and open logs by **Notepad++** or other editor software.

Delete system logs

1. Log in to PBX web portal, go to **Maintenance > System Logs**.
2. To delete a system log, click  beside the desired log.
3. To bulk delete system logs, select the checkboxes of the desired logs, click **Delete**.

CDR and Reports

CDR

Call Detail Record (CDR) Overview

The Call Detail Record (CDR) feature provides information about calls over Yeastar P-Series Cloud Edition. This topic describes parameters and auto cleanup of CDR.

CDR parameters

A CDR contains the following information:

Item	Description
ID	A unique identifier for each call.
Time	When the call was made or received.
Call From	The number or the name of the caller.
Call To	The number or the name of the callee.
Call Duration	The time between the call started and the call ended.
Ring Duration	The time between the call started and the call answered.
Talk Duration	The time between the call answered and the call ended.
Status	Call status. • **ANSWERED** • **NO ANSWER** • **BUSY** • **FAILED** • **VOICEMAIL**
Reason	The reason why the call was ended.
Source Trunk	The call was received via which trunk.
Destination Trunk	The call was sent out via which trunk.
Communication Type	Communication Type. • **Internal** • **Outbound** • **Inbound**

Item	Description
DID/DDI	The phone number that the caller dialed.
Outbound Caller ID	The phone number that was displayed on the callee's phone.
Caller IP Address	The IP address of the caller's device.
PIN Code	The PIN code entered when making a call via a restricted outbound route.
Recording File	The call recording file.

CDR auto cleanup

By default, when the number of call logs reaches 200,000, the system will automatically delete the oldest call logs (relevant recordings are retained.). You can change the maximum value, or you can also set the maximum preservation days. For more information, see Auto Cleanup Settings.

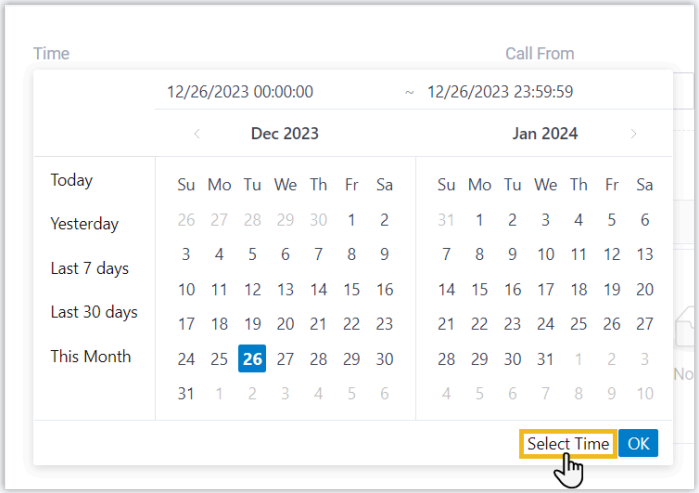
View and Manage CDR

This topic describes how to view, download, and delete CDR.

View CDR

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. **Optional:** Set the basic filter criteria.

Filter	Description
Time	Set the start date and the end date. **Note:** To specify a time period of a day, click **Select Time** to set the start time and the end time.

Filter	Description
	
Call From	Enter the caller's number or name.
Call To	Enter the callee's number or name. **Tip:** To swap the callee for the caller, click ⇆.
Status	Select a call status. • **All** • **ANSWERED** • **NO ANSWER** • **BUSY** • **FAILED** • **VOICEMAIL**

3. **Optional:** Set the advanced filter criteria.

   a. Click .

   b. On the **Filter** page, set the advanced criteria.

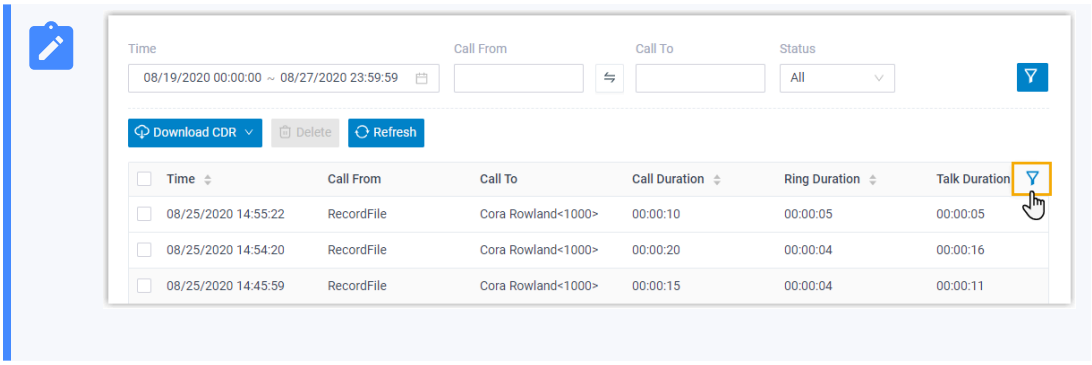
Option	Description
Queue	Select a queue. The system only queries the queue's calls.
Extension Group	Select an extension group. The system only queries the group members' calls.
Ring Duration	Enter how long the callee's phone rang before the call was answered.

Option	Description
	Note: Only numbers, -, =,<, <=, >, and >= are allowed.
Talk Duration	Enter the time between the call was answered and the call was ended. **Note:** Only numbers, -, =,<, <=, >, and >= are allowed.
Status	Select call status. • **All** • **ANSWERED** • **NO ANSWER** • **BUSY** • **FAILED** • **VOICEMAIL**
Communication Type	Select communication type. • **All** • **Internal** • **Outbound** • **Inbound**
ID	Enter the unique ID for a call.
Trunk	Select a trunk. The system only queries calls that are sent or received via the trunk.
Enable Number Fuzzy Search	Set whether to search for the fuzzy equivalent for the phone number.
PIN Code	Enter an existing PIN code. The system only queries the calls using this PIN code.

c. Scroll up to click ✕ to close the window.
The filtered call logs are displayed on the page.

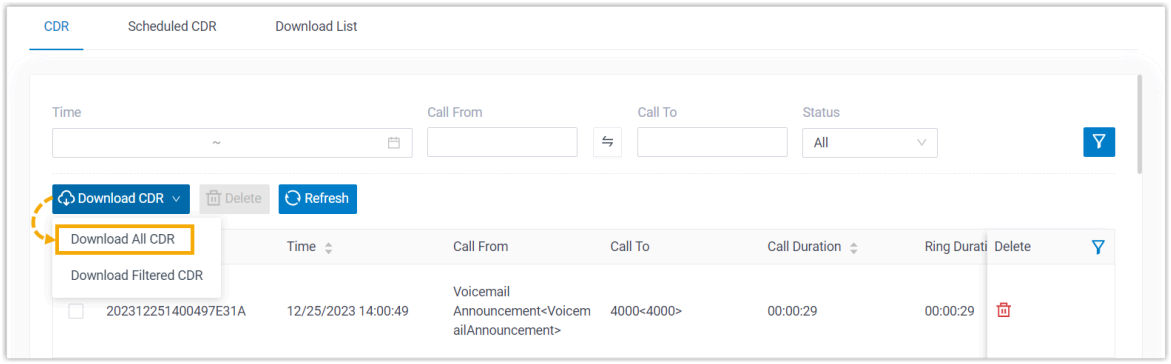
Note:

You can click to decide which item will be displayed.

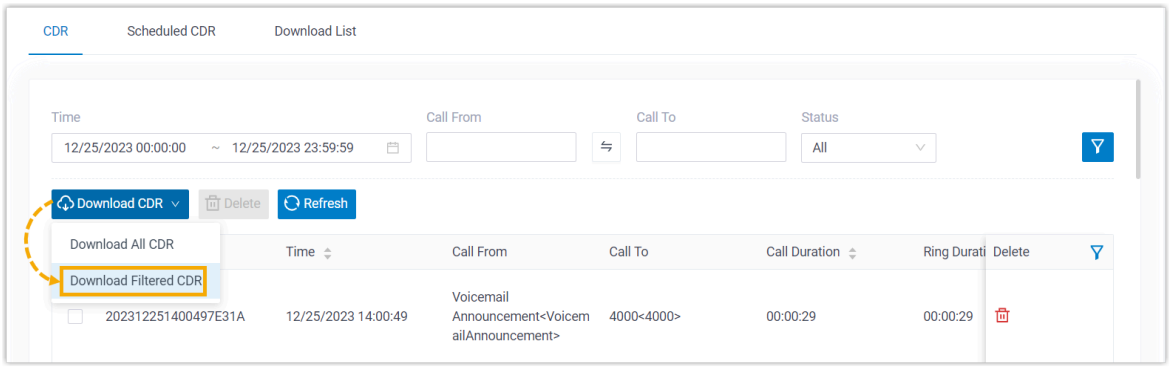


Download CDR

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. To download all the call logs, select **Download All CDR** from the drop-down list of **Download CDR**.



3. To download the filtered call logs, set the filter criteria and select **Download Filtered CDR** from the drop-down list of **Download CDR**.



CDR are exported to a CSV file.

Delete CDR

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. **Optional:** Filter call logs.
3. Select the checkboxes of the desired call logs, click **Delete** and **OK**.

> **Important:**
> The relevant recording files will also be deleted.

Both call logs and recording files are deleted.

Scheduled CDR

Schedule CDR to Email

Yeastar P-Series Cloud Edition allows you to schedule CDR to be sent to specific recipients' mailboxes at the specified time. The recipients can access the CDR via a link attached in the email and download it as a CSV, XLS, or PDF file.

Restrictions

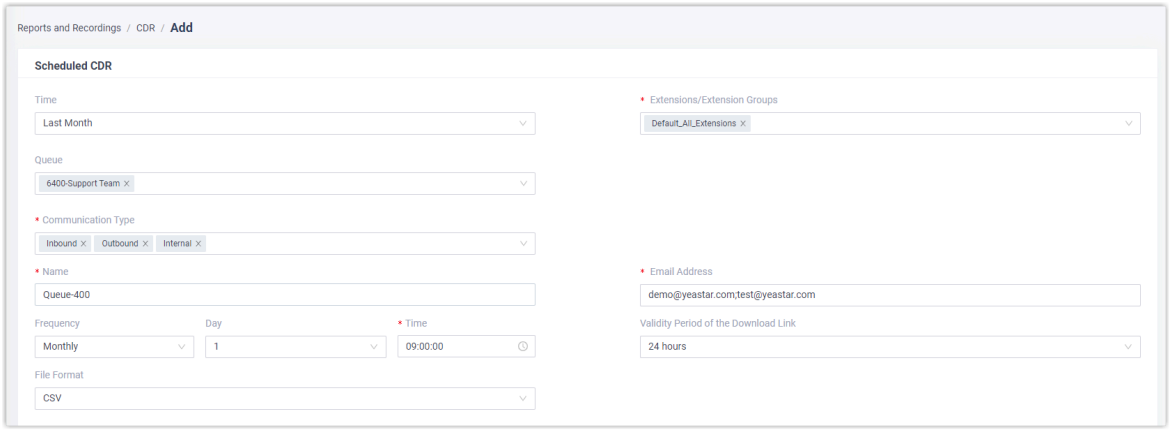
- Only super administrator and the authorized users can schedule CDR to email.
- The maximum number of CDR that can be downloaded at one time varies depending on the file format of the downloaded CDR.
  - **CSV**: 10,000
  - **XLS**: 10,000
  - **PDF**: 5,000

Prerequisites

Make sure email server works.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. Under the **Scheduled CDR** tab, click **Add**.
3. Schedule a task to send CDR via email.



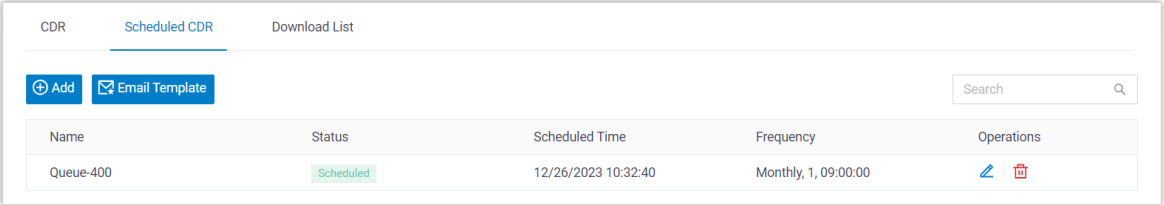
Setting	Description
Time	Select a time range to filter CDR.
Extensions/Extension Groups	Select extension(s) or extension group(s) to filter CDR.
Organization	Select department(s) to filter CDR.
Queue	Select call queue(s) to filter CDR.
Communication Type	Select communication type(s) to filter CDR.
Name	Enter a name to help you identify the task.
Email Address	Enter recipients' email addresses. **Note:** You can set up to 10 email addresses; Separate multiple email addresses by semicolon ;.
Frequency	Set how often to send CDR. • **Once**: If selected, the system will send the CDR immediately after you save the task. • **Daily**: If selected, choose a specific time from the drop-down list. The system will send the CDR at the selected time each day. • **Weekly**: If selected, choose a specific day of the week and select a specific time from the drop-down list. The system will send the CDR at the selected day and time each week. • **Monthly**: If selected, choose a specific day of the month and select a specific time from the drop-down list. The system will send the CDR on the selected day and time each month.

Setting	Description
Validity Period of the Download Link	Set the validity period of the download link for the scheduled CDR. After the link expires, the recipients can NOT access and download the CDR via the link.
File Format	Set in which file format the CDR can be downloaded. • **CSV** • **XLS** • **PDF**

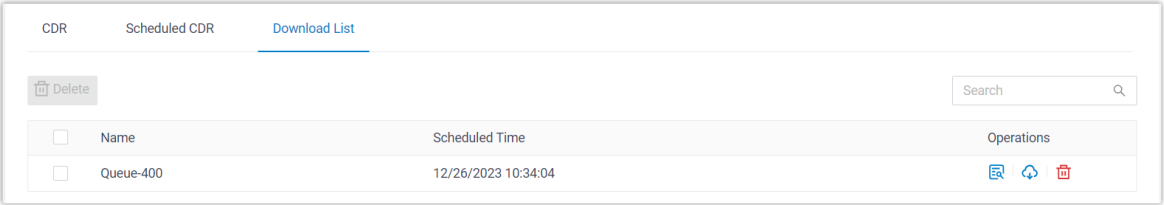
4. Click **Save**.

Result

- The task is scheduled and displayed in the list.



- When it comes to the scheduled time, the specified CDR will be automatically sent to the specified recipients' mailboxes; You can also check the CDR in the **Download List** tab, either by clicking to preview the CDR online, or by clicking to download the CDR for offline viewing.



Related information

Manage Scheduled CDR Tasks
Customize Email Template for Scheduled CDR

Manage Scheduled CDR Tasks

This topic describes how to edit and delete scheduled CDR tasks.

Edit a scheduled CDR task

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. Under the **Scheduled CDR** tab, click beside the desired task.
3. Edit the task according to your needs.
4. Click **Save**.

Delete a scheduled CDR task

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. Under the **Scheduled CDR** tab, do as follows:
   a. Click beside the desired task.
   b. In the pop-up window, click **OK**.

Download Scheduled CDR on PBX Web Portal

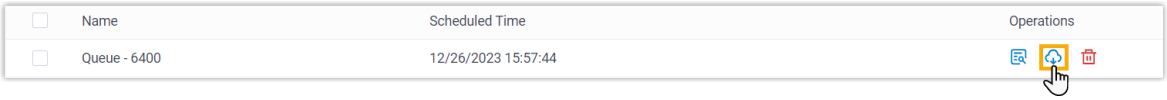
After the system sends scheduled CDR to recipients' mailboxes, the recipients can preview and download the CDR via an attached link, while you can also view and download the CDR on PBX web portal. This topic describes how to download scheduled CDR on PBX web portal.

Prerequisites

A scheduled CDR task has been performed successfully.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. Under **Download List** tab, click beside the desired scheduled CDR task.



Result

The CDR is downloaded to your computer in the pre-defined file format.

Customize Email Template for Scheduled CDR

This topic describes how to customize the email template for scheduled CDR.

Background information

By default, Yeastar P-Series Cloud Edition sends scheduled CDR in the pre-defined language and in a default email template.

The language is what you have set in system email template, and the default email template contains the following information:

- A download link for CDR.
- The validity period of the download link.
- System information, including PBX name and PBX serial number.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > CDR**.
2. Under the **Scheduled CDR** tab, click **Email Template**.
3. Configure template settings.
    a. In the **Template** drop-down list, select **Custom**.
    b. Edit email subject and content according to your needs.
    c. Click **Save**.

Result

The PBX will send scheduled CDR using the customized email template.

Call Report

Call Reports Overview

Yeastar P-Series Cloud Edition provides intuitive visual call reports, which allow you to check call statistics of different objects, such as extensions, trunks, queues, ring groups, etc. This topic describes category of call reports, and methods of getting an instant or a scheduled call report.

Types of Call Reports

Yeastar P-Series Cloud Edition supports the following types of call reports:

- Extension Call Statistics
- Extension Call Activity
- Trunk Activity
- DID/Outbound Caller ID Activity
- IVR Report
- Queue AVG Waiting & Talking Time
- Queue Performance
- Queue Performance Activity
- Queue Callback Summary
- Queue Callback Activity
- Satisfaction Survey
- Satisfaction Survey Details
- Agent Login Activity
- Agent Pause Activity
- Agent Missed Call Activity
- Agent Performance
- Agent Call Summary
- Ring Group Statistics
- Extension Call Accounting
- Extension Call Accounting Details

Methods of getting a call report

Yeastar P-Series Cloud Edition allows you to have an instant search and view of call reports on PBX web portal, or schedule call reports to be sent to your mailbox at the specified time and download the reports to your local device.

For more information about searching and viewing call reports on PBX web portal, see View Call Reports.

For more information about scheduling call reports, see Schedule Call Reports.

Call Reports

View Call Reports

This topic describes how to view call reports on Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports > Call Reports**.
2. Set search criteria.
    a. In the **Report Type** drop-down list, select the desired report.
    b. Set a time period that the report covers.
    c. Set one or more objects that you want to query.

Result

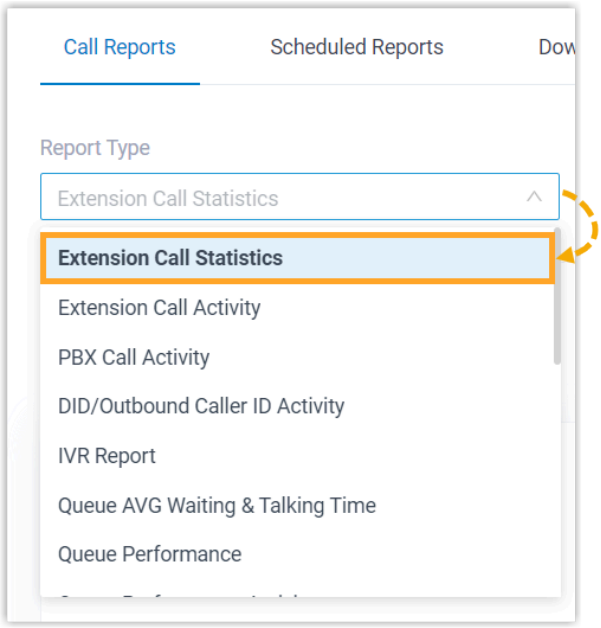
Relevant call statistics are displayed on the page.

Extension Call Statistics Report

Extension Call Statistics report provides a quick overview of the number of calls that have been made and received by extension(s), and can be broken down to specific extensions, groups of extensions, or organizations of extensions. This topic introduces how to access the report and explains the key metrics in detail.

Access Extension Call Statistics report

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. In the **Report Type** drop-down list, select **Extension Call Statistics**.

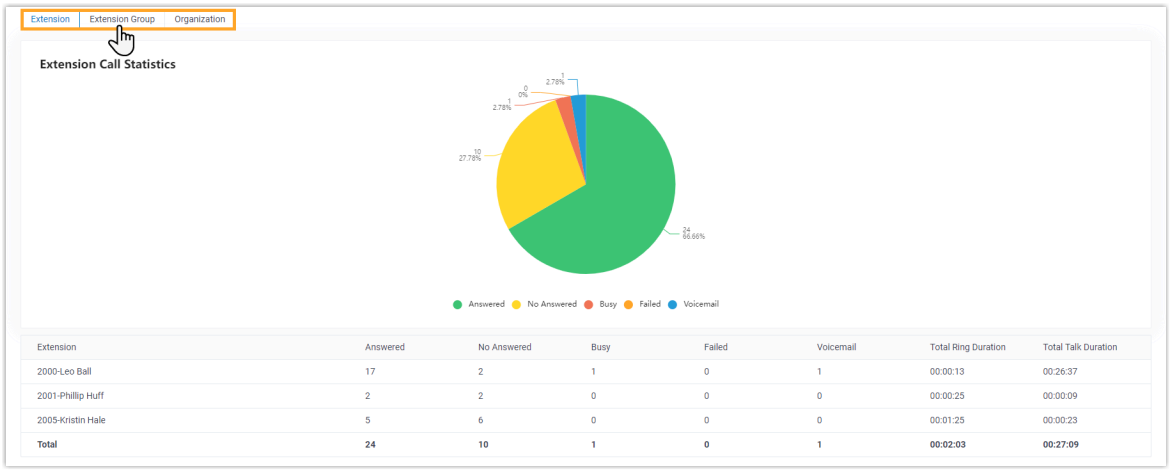


3. Filter data by system time, extension(s), extension group(s), organization(s), or communication type.

> **Note:**
> **Organization** filter is displayed only when you enable **Organization Management** feature (Path: **PBX Settings > Preferences**).



A report that meets the filter criteria is displayed on the page, as shown below.



> **Note:**
> - You can click the tab at the top-left corner to switch between the breakdown reports.
> - If you filter data by organization(s), the report data will include the selected organization and its subordinate departments.

Report details

We take the above report as an example to introduce the key metrics for **Extension Call Statistics** report.

Metric	Description
Answered	The total number of calls that the extension(s) answered.
No Answered	The total number of calls that were routed to the designated destination when the extension(s) didn't answer the calls.

Metric	Description
Busy	The total number of calls that were routed to the designated destination when the extension(s) was busy.
Failed	The total number of calls that were failed to be made by the extension(s).
Voicemail	The total number of voicemails that the extension(s) received.
Total Ring Duration	The total time between calls started and calls answered. **Note:** This metric is displayed only when you set **Communication Type** filter to **All**, **Inbound**, **Outbound**, or **Internal**.
Total Talk Duration	The total time between calls answered and calls ended.
Total (displayed in row)	The sum of all the values in each column. **Note:** • This metric is only available for extension and extension group. • If you select multiple extension groups and one of the extensions belongs to two or more extension groups, the call statistics for the extension will be calculated more than once.
Total (displayed in column)	The total number of calls for each communication type. **Note:** This metric is displayed only when you set **Communication Type** filter to **Inbound/Outbound**.

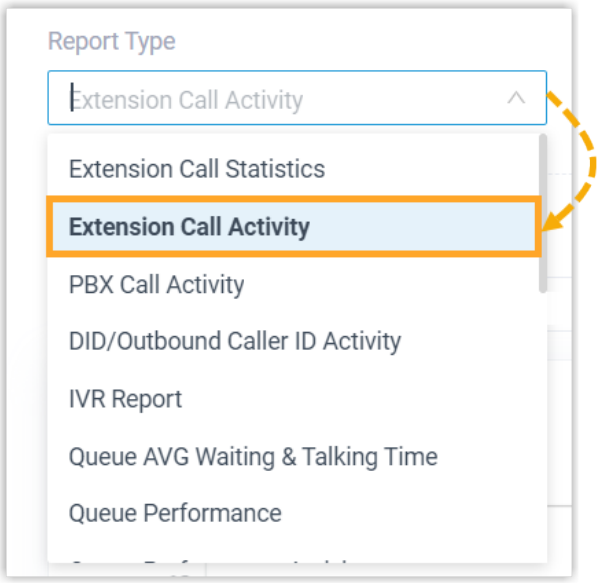
Extension Call Activity Report

Extension Call Activity report provides granular insights into the hourly, daily, and monthly breakdown of the number of calls that have been made or received by extension(s), and can be broken down to specific extensions, groups of extensions, or organizations of extensions. This topic introduces how to access the report and explains the key metrics in detail.

Access Extension Call Activity report

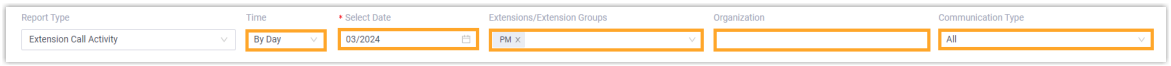
1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.

2. In the **Report Type** drop-down list, select **Extension Call Activity**.

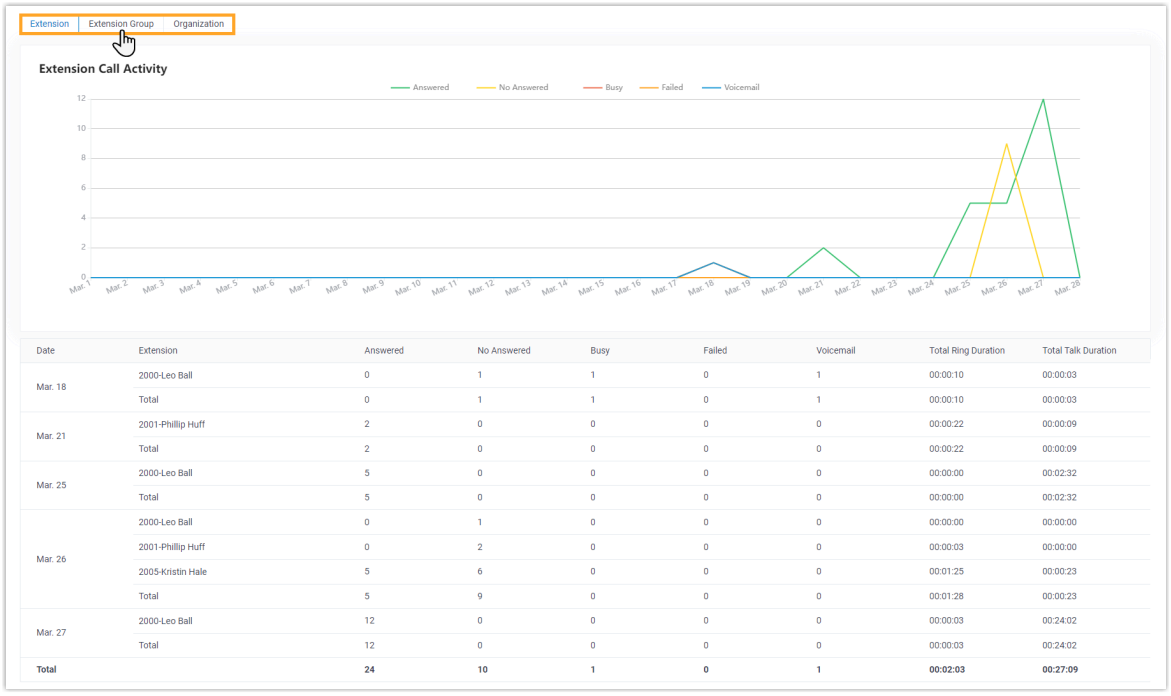


3. Filter data by system date and time, extension(s), extension group(s), organization(s), or communication type.

> **Note:**
> **Organization** filter is displayed only when you enable **Organization Management** feature (Path: **PBX Settings > Preferences**).



A report that meets the filter criteria is displayed on the page, as shown below.



Date	Extension	Answered	No Answered	Busy	Failed	Voicemail	Total Ring Duration	Total Talk Duration
Mar. 18	2000-Leo Ball	0	1	1	0	1	00:00:10	00:00:03
	Total	0	1	1	0	1	00:00:10	00:00:03
Mar. 21	2001-Phillip Huff	2	0	0	0	0	00:00:22	00:00:09
	Total	2	0	0	0	0	00:00:22	00:00:09
Mar. 25	2000-Leo Ball	5	0	0	0	0	00:00:00	00:02:32
	Total	5	0	0	0	0	00:00:00	00:02:32
Mar. 26	2000-Leo Ball	0	1	0	0	0	00:00:00	00:00:00
	2001-Phillip Huff	0	2	0	0	0	00:00:03	00:00:00
	2005-Kristin Hale	5	6	0	0	0	00:01:25	00:00:23
	Total	5	9	0	0	0	00:01:28	00:00:23
Mar. 27	2000-Leo Ball	12	0	0	0	0	00:00:03	00:24:02
	Total	12	0	0	0	0	00:00:03	00:24:02
Total		**24**	**10**	**1**	**0**	**1**	**00:02:03**	**00:27:09**

Note:

- You can click the tab at the top-left corner to switch between the break-down reports.
- If you filter data by organization(s), the report data will include the selected organization and its subordinate departments.

Report details

We take the above report as an example to introduce the key metrics for **Extension Call Activity** report.

Metric	Description
Answered	The total number of calls that the extension(s) answered.
No Answered	The total number of calls that were routed to the designated destination when the extension(s) didn't answer the calls.
Busy	The total number of calls that were routed to the designated destination when the extension(s) was busy.
Failed	The total number of calls that were failed to be made by the extension(s).
Voicemail	The total number of voicemails that the extension(s) received.
Total Ring Duration	The total time between calls started and calls answered.

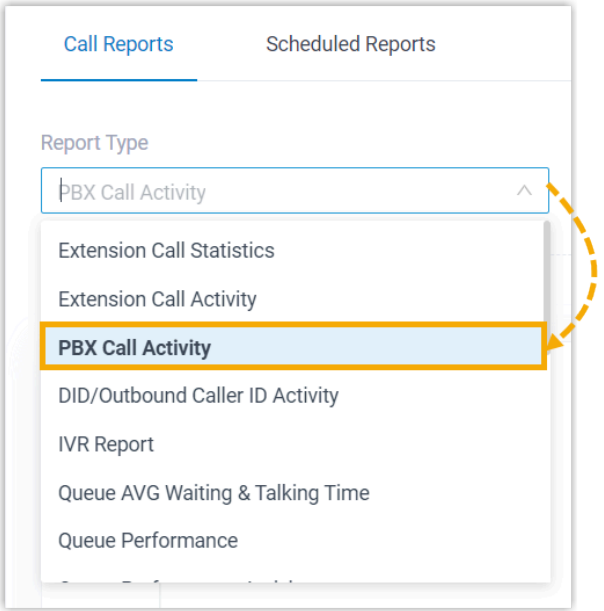
Metric	Description
Total Talk Duration	The total time between calls answered and calls ended.
Total (displayed in row)	The sum of all the values in each column. **Note:** • This metric is only available for extension and extension group. • If you select multiple extension groups and one of the extensions belongs to two or more extension groups, the call statistics for the extension will be calculated more than once.

PBX Call Activity Report

PBX Call Activity report provides granular insights into the hourly, daily, and monthly break-down of internal calls and external calls. This topic introduces how to access the report and explains the key metrics in detail.

Access PBX Call Activity report

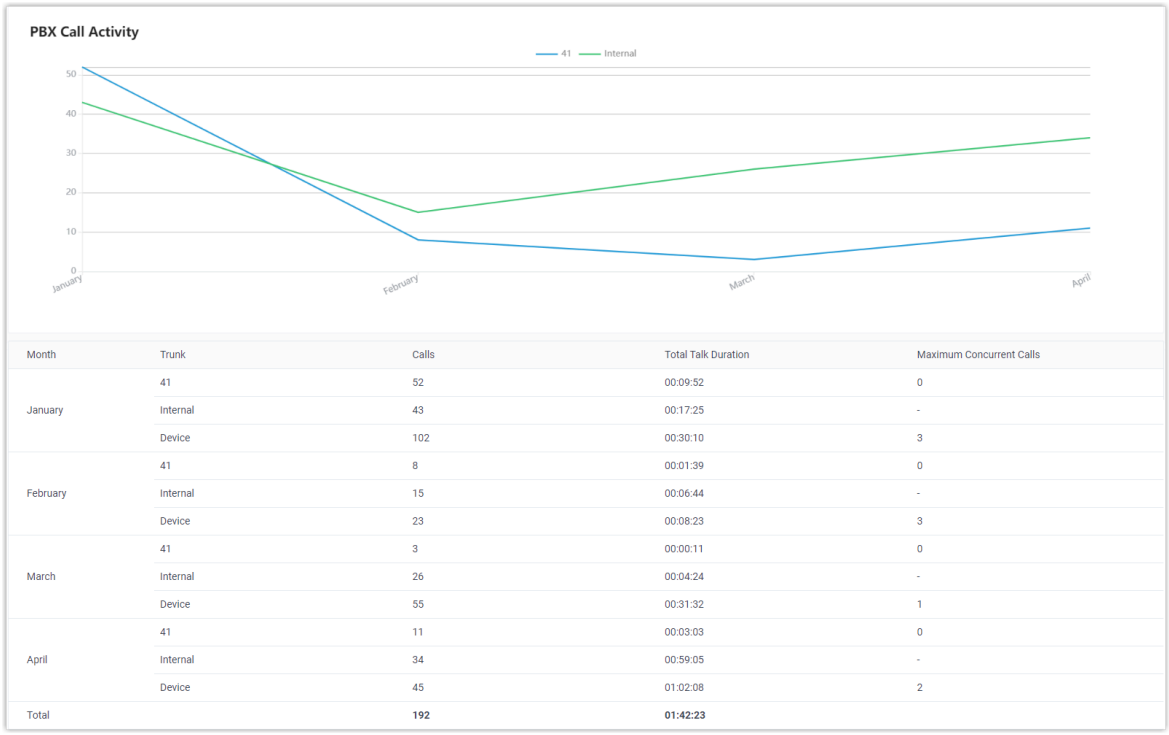
1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. In the **Report Type** drop-down list, select **PBX Call Activity**.



3. Filter data by system time, trunks, call type, or communication type.



A report that meets the filter criteria is displayed on the page, as shown below.



Month	Trunk	Calls	Total Talk Duration	Maximum Concurrent Calls
	41	52	00:09:52	0
January	Internal	43	00:17:25	-
	Device	102	00:30:10	3
	41	8	00:01:39	0
February	Internal	15	00:06:44	-
	Device	23	00:08:23	3
	41	3	00:00:11	0
March	Internal	26	00:04:24	-
	Device	55	00:31:32	1
	41	11	00:03:03	0
April	Internal	34	00:59:05	-
	Device	45	01:02:08	2
Total		**192**	**01:42:23**	

Report details

We take the above report as an example to introduce the key metrics for **PBX Call Activity** report.

Metric	Description
Trunk	Trunk name. • *{trunk_name}*: Calls that are made or received via trunks. • **Internal**: Calls that are made between extensions. • **Device**: Calls of the entire system (including internal calls, inbound and outbound calls on all the trunks). **Note:** This value is displayed only when **Communication Type** filter is set to **All**.
Calls	The total number of calls that were made or received.

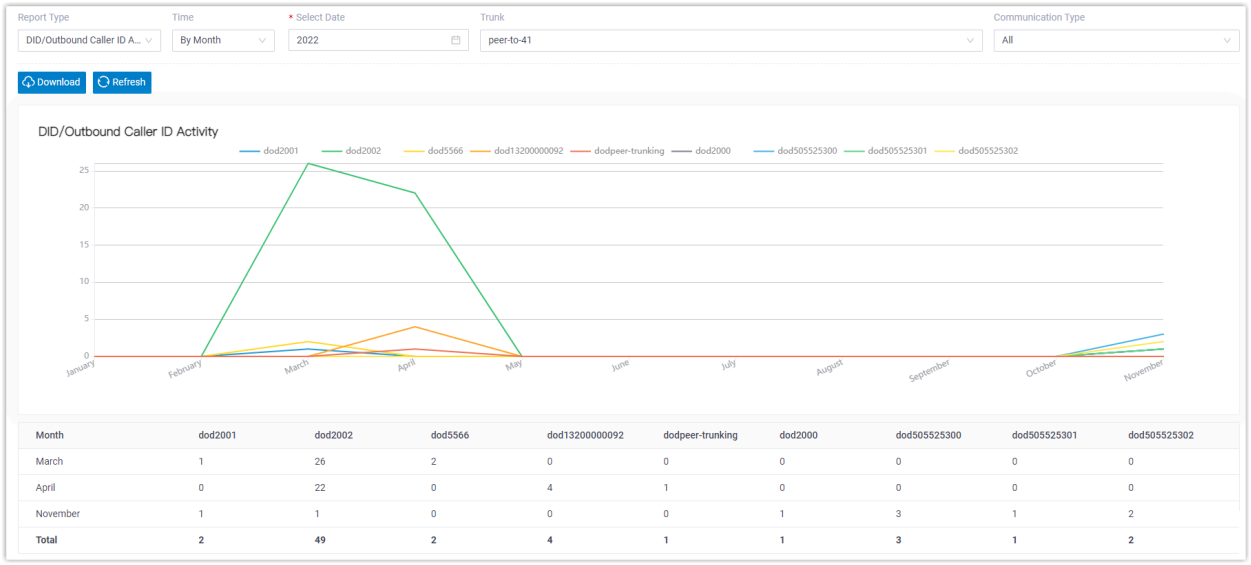
Metric	Description
Total Talk Duration	The total time between calls answered and calls ended.
Maximum Concurrent Calls	The maximum number of concurrent calls within the time frame. **Note:** • Calls between extensions and the calls received from WebRTC trunks do NOT support the calculation of maximum concurrent calls. • For **Trunk** displayed as **Device**, it calculates the maximum concurrent calls for the entire phone system.
Total (displayed in row)	Total number of calls and total talk duration. **Note:** This shows the sum of the data for internal calls and the selected trunks, excluding the data for the entire phone system (namely the data for **Trunk** displayed as **Device**).

DID/Outbound Caller ID Activity Report

DID/Outbound Caller ID Activity report is a summary report displayed in line graph, which allows you to track changes of DID/Outbound Caller ID activity by hour, by date, or by month.

Report example

The following report shows monthly DID/Outbound Caller ID statistics of the trunk **peer-to-41** on 2022.



Month	dod2001	dod2002	dod5566	dod13200000092	dodpeer-trunking	dod2000	dod505525300	dod505525301	dod505525302
March	1	26	2	0	0	0	0	0	0
April	0	22	0	4	1	0	0	0	0
November	1	1	0	0	0	1	3	1	2
Total	**2**	**49**	**2**	**4**	**1**	**1**	**3**	**1**	**2**

Ring Group Statistics Report

Ring Group Statistics report allows you to query the number of received and answered calls for a specific ring group, thus helping you evaluate performance of members within the ring group.

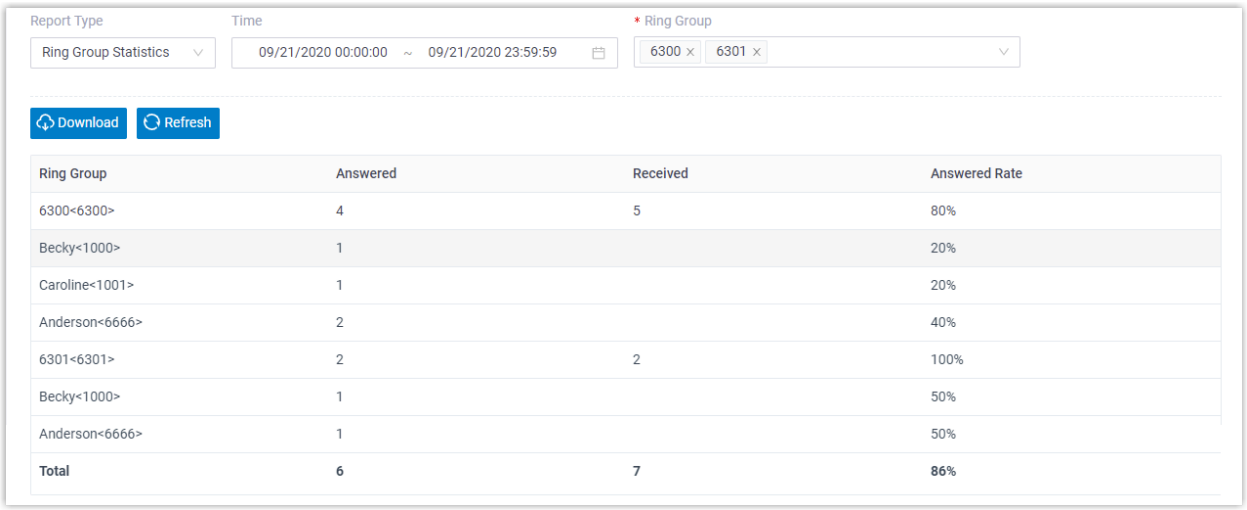
Report details

The following table lists the related parameters for Ring Group Statistics report.

Parameter	Description
Answered	The total number of calls that were answered.
Received	The total number of calls that were received.
Answered Rate	The answered rate for the ring group or for each group member.

Report example

The following report shows call statistics of ring group **6300** and **6301** on 09/21/2020.



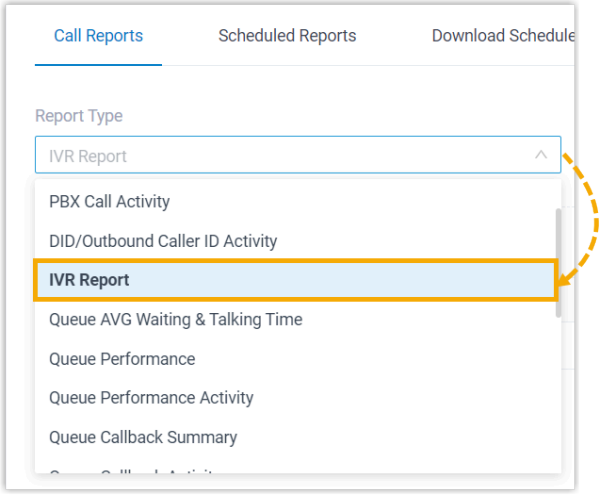
Ring Group	Answered	Received	Answered Rate
6300<6300>	4	5	80%
Becky<1000>	1		20%
Caroline<1001>	1		20%
Anderson<6666>	2		40%
6301<6301>	2	2	100%
Becky<1000>	1		50%
Anderson<6666>	1		50%
Total	**6**	**7**	**86%**

IVR Report

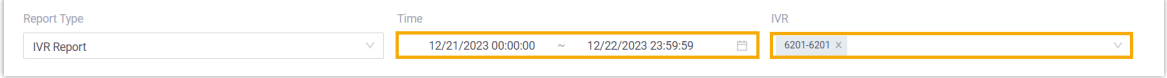
IVR report provides a quick overview of the keypress events for IVR as well as granular insights into the associated IVR calls. This topic introduces how to access the report and explains the key metrics in detail.

Access IVR Report

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. In the **Report Type** drop-down list, select **IVR Report**.



3. Filter data by system time and IVR(s).



A report that meets the filter criteria is displayed on the page, as shown below.



IVR	Press0	Press1	Press2	Press3	Press4	Press5	Press6	Press7	Press8	Press9	Press#	Press*	Response Timeout	Invalid Input Destination	Details
6201-6201	1	0	1	0	0	0	0	0	1	0	0	0	1	1	

Report details

We take the above report as an example to introduce the key metrics for **IVR** report.

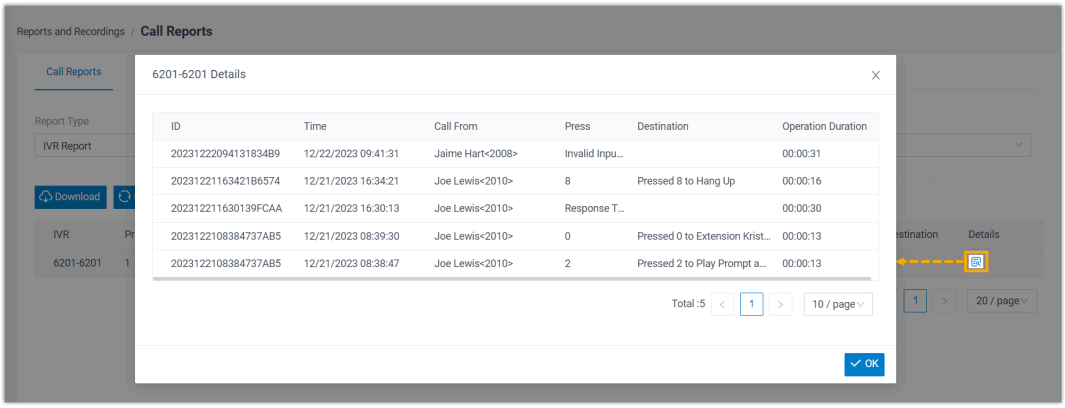
Metric for IVR



IVR	Press0	Press1	Press2	Press3	Press4	Press5	Press6	Press7	Press8	Press9	Press#	Press*	Response Timeout	Invalid Input Destination	Details
6201-6201	1	0	1	0	0	0	0	0	1	0	0	0	1	1	

Metric	Description
Press*key_pressed*	The number of times that the key has been pressed.
Response Timeout	The number of times that IVR calls were routed to the Timeout destination. **Note:** IVR calls will be routed to response timeout destination if no key was pressed after reaching the maximum **Prompt Repeat Count**.
Invalid Input Destination	The number of times that IVR calls were routed to the invalid input destination. **Note:** IVR calls will be routed to invalid input destination if callers repeatedly entered a DTMF digit that is not defined for the IVR after reaching the maximum **Prompt Repeat Count**.

Metric for IVR calls



Metric	Description
ID	A unique ID for the call.
Time	When the call was received.
Call From	The number and the name of the caller.
Press	The keypress event.
Destination	The destination of the call.
Operation Duration	The time between the caller called into the IVR and exited the IVR.

Scheduled Reports

Schedule Call Reports

This topic describes how to schedule a call report to be sent to specific recipients' mailboxes at the specified time.

Background information

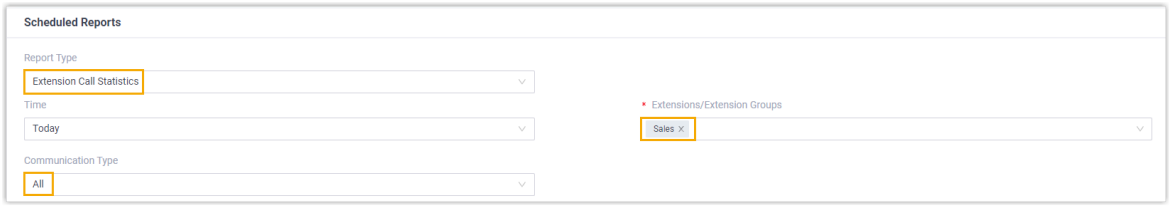
A scheduled call report is a diagram containing call statistics for the selected objects within a specific time frame. It automatically runs at a pre-defined frequency and is emailed to a specific address as a link and can be downloaded in CSV, XLS, or PDF.

Prerequisites

- Make sure email server works.
- Customize Email Template for Scheduled Reports.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Under **Scheduled Reports** tab, click **Add Report**.
3. In the **Report Type** drop-down list, choose the desired report, and select one or more objects that you want to query.



Note:
Descriptions for specific parameters:

- **Communication Type**: Select a communication type from **Inbound**, **Outbound**, **Internal**, **Inbound/Outbound**, or **All**. You can use the parameter to filter call statistics in the following reports:
  - **Extension Call Statistics**
  - **Extension Call Activity**
  - **Trunk Activity**
  - **DID/Outbound Caller ID Activity**
- **Short Abandoned Calls**: Set a time. Calls abandoned within the specified time will not be included in report. You can use the parameter to filter call statistics in the following reports:
  - **Queue Performance**
  - **Queue Performance Activity**
  - **Agent Missed Call Activity**
  - **Agent Performance**
- **Organization**: Select one or more departments. Calls of the department and its sub-departments will be included in the report.

  Note:
  This option is available only when you enable the **Organization Management** feature.
  You can use the parameter to filter call statistics in the following reports:
  - **Extension Call Statistics**

- **Extension Call Activity**
- **Extension Call Accounting**
- **Extension Call Accounting Details**

4. Schedule the report.
   - **Time**: Set a time frame that the desired report covers.
   - **Report Name**: Enter a name to help you identify it.
   - **Email Address**: Enter recipients' email addresses, separated by semicolon `;`.

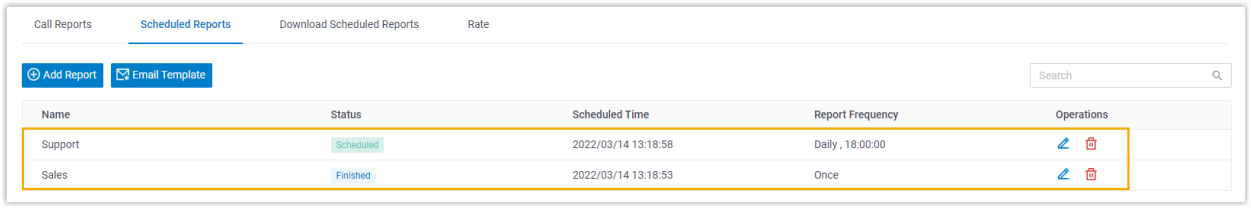
     Note:
     You can set up to 10 email addresses.

     The report will be sent to the email address at the specified time.
   - **Report Frequency**: Set how often to send the report.
     - **Once**: If you choose the option, the system sends the report immediately after you save the setting.
     - **Daily**: If you choose the option, select a time from the drop-down list. The system sends the report at this time of the day.
     - **Weekly**: If you choose the option, choose a day of week and select a time from the drop-down list. The system sends the report at this time of the week.
     - **Monthly**: If you choose the option, choose a day and select a time from the drop-down list. The system sends the report on this day and time of the month.
   - **Validity Period of the Download Link**: Set the validity period of the download link for the scheduled report.

     After the link expires, you can NOT download the report via the link.
   - **Format**: Set in which format the report can be downloaded.
     - **CSV**
     - **XLS**
     - **PDF**
5. Click **Save**.

Result

On **Scheduled Reports** list, check status of the scheduled call report.



- **Finished**: The one-off call report was sent to the recipients' email addresses.
- **Scheduled**: The call report is scheduled and valid. The system will send the report to the recipients' email addresses at the specified time.
- **Paused**: The scheduled call report is on hold because the Call Center Service expired. To renew it, go to Yeastar P-Series Enterprise Plan or Ultimate Plan.

Manage Scheduled Reports

This topic describes how to edit and delete scheduled reports.

Edit scheduled reports

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**, click **Scheduled Reports**.
2. Select the desired report, click .
3. Edit the scheduled report according to your needs.
4. Click **Save**.

Delete scheduled reports

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**, click **Scheduled Reports**.
2. Select the desired report, click .
3. In the pop-up dialog box, click **OK**.

Download Scheduled Reports on Web Interface

After the system sends scheduled reports to recipients' mailboxes, the recipients can download reports via attached links and system administrator can view and download reports on PBX web portal. This topic describes how to download scheduled reports on PBX web portal.

Prerequisites

A scheduled report was sent out.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**, click **Download Scheduled Reports** tab.
2. Select the desired call report, click  .

   The report contains a snapshot of data for the time frame you have selected.
3. At the top-right corner, click **Download**.

Result

The report is downloaded to your computer in the pre-defined format.

Customize Email Template for Scheduled Reports

This topic describes how to customize email template for scheduled reports.

Background information

By default, Yeastar P-Series Cloud Edition sends scheduled call reports in the pre-defined language and email template.

The language is what you have set in system email template, and the email template contains the following information:

- A download link for call report.
- Soft reminder of the download link.
  - The link is valid for 24 hours.
- System information, including PBX name, PBX serial number.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports > Scheduled Reports**.
2. Click **Email Template**.
3. Configure template settings.
   a. In the **Template** drop-down list, select **Custom**.

b. Edit email subject and content according to your needs.
c. Click **Save**.

Result

The PBX will use the email template to send scheduled reports.

Call Accounting

Call Accounting Overview

The Call Accounting feature collects and records telecom usage, as well as estimating the expenses incurred based on the destination number and call duration, allowing you to get an eagle-eye overview of the telephone activity and cost of all employees.

Scenario

In customer service businesses, many employees need to call potential customers via National Direct Dialing (NDD) or International Direct Dialing (IDD). Leaving such outbound calls unattended makes the way for telecommunication misuse, and eventually causes financial losses. These companies need a management tool to track telephone activity of employees, so as to keep telecom usage and expenses in control.

Highlights

Collect and record call statistics

Monitor, record, and track individual and department calls. Based on the historical and real-time call logs, you can measure and analyze employees' performance and identify telephone misuse and abuse.

Flexible call rate settings

Apply different call rate rules to local calls, long distance calls, and international calls. In this way, you can monitor the billing statistics for each type of call, and estimate call charges before telecommunications provider sends the bill.

Insightful call accounting reports

Get dedicated call accounting reports based on individuals or departments. Detailed information about the total number of calls, total call duration, average call duration, and total billing gives you deeper insights into the calling patterns and activity of employees.

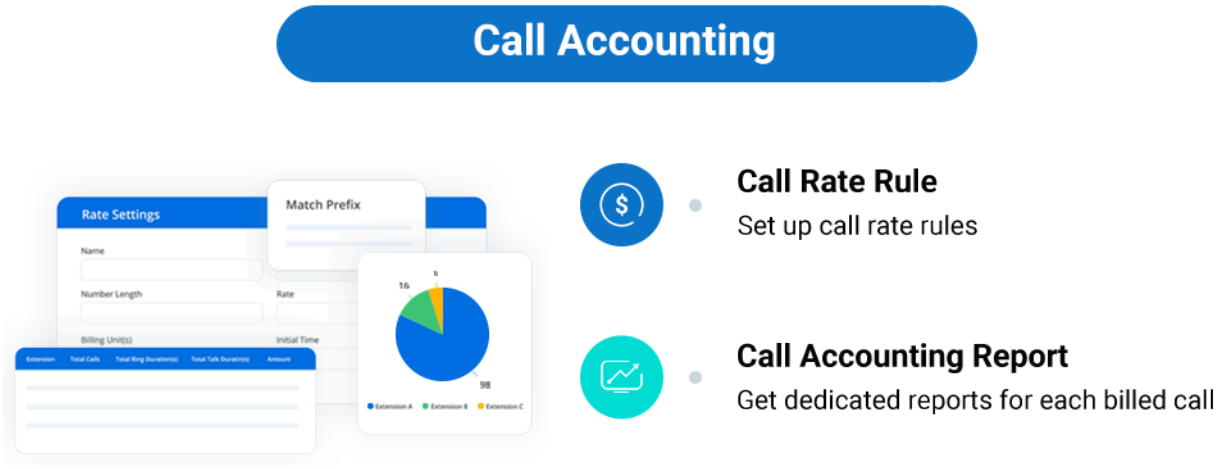
Improve telecom budget planning

Tracking and analyzing telecom usage by individuals or departments help you manage and control telecommunication expenses, making the budgeting more accurate and efficient.

A quick glance at Call Accounting

We provide a thumbnail for you to glance at the WebGUI and get to know the workflow of call accounting.

> **Tip:**
> You can click on the text on the right of the thumbnail to quickly redirect to the corresponding topic.



1. Add a Call Rate Rule
2. Extension Call Accounting Report

Call Rate

Add a Call Rate Rule

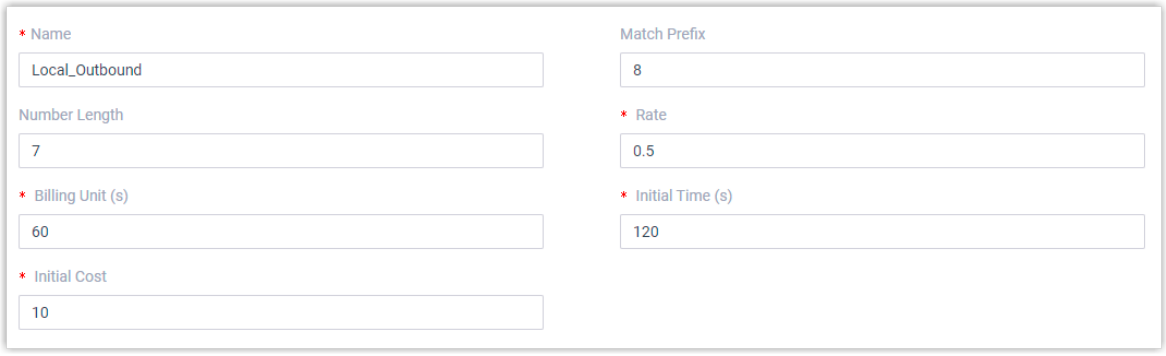
To monitor telecommunication costs for groups and individuals and prevent from potential fraudulent use of resources, you can add a call rate rule for outbound calls, such as local, long-distance, or international calls.

Restrictions

- A maximum number of 30,000 call rate rules is supported on Yeastar P-Series Cloud Edition.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Under **Rate** tab, click **Add**.
3. Set up a call rate rule:



- **Name**: Enter a name to help you identify the call rate rule. For example, enter `Local_Outbound`.
- **Match Prefix**: Optional. Define the dialing prefix to match the call rate rule.

  > **Note:**
  > Only outbound calls matching the dialing prefix will apply this rate.

- **Number Length**: Optional. Define the length of the dialed number to match the call rate rule.

  > **Note:**
  > Only the dialed number whose length is shorter than or equal to the value will match this rule.

- **Rate**: Enter a call rate. After the initial time, each billing unit will be charged with this rate.
- **Billing Unit (s)**: Define the time increment (in seconds) that will be used to calculate the fee for a call after the initial time. The default value is 60 seconds.

  For example, set **Rate** to 0.5 and **Billing Unit** to 60 seconds. In this way, the fee for a call will increase by 0.5 every 60 seconds.
- **Initial Time (s)**: Define the initial period of time (in seconds) during which the call will be charged with the initial cost.
- **Initial Cost**: Define the fixed cost incurred over the preset initial time.

For example, set **Initial Time** to 120 seconds and **Initial Cost** to 2. In this case, it costs 2 for the call within 2 minutes. After 2 minutes, the call will be charged with the preset rate.

4. Click **Save** and **Apply**.

Result

- The call rate rule is created. Any outbound calls matching the rule will be charged with the call rate as the following formula from now on.

```
Amount = Initial Cost + [(Talk Duration - Initial Time) / Billing Unit] * Rate
```

> **Note:**
> If the value to be multiplied by rate has decimals, it will be rounded upwards to the nearest integer.

- If you create more than one call rate rules in the list, the system will match outbound calls with these rules from the top down. You can click ⤒ ^ v ⤓ to adjust the priority.

What to do next

You can check the call accounting fee for extensions, extension groups, or departments in call reports.

For more information, see Extension Call Accounting Report and Extension Call Accounting Details Report.

Manage Call Rate Rules

This topic describes how to edit and delete call rate rules.

Adjust priority of call rate rules

When users make outbound calls, the system will match the outbound calls with call rate rules in the list from the top down, and use the first matched rule to charge the outbound calls. You can adjust the priority of call rate rules according to your needs.

1. Log in to PBX web portal, go to **Reports and Recording Files > Call Reports**.
2. Under **Rate** tab, click ⤒ ^ v ⤓ to adjust the priority of call rate rules.

3. Click **Apply**.

Edit a call rate rule

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Under **Rate** tab, click  beside a call rate rule.
3. Edit the call rate rule.

> **Important:**
> Modifying the rate would affect the newly generated call billing statistics.

4. Click **Save** and **Apply**.

Delete call rate rules

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Click **Rate** tab.
3. To delete a call rate rule, do as follows:
   a. On the right of a desired call rate rule, click  .
   b. In the pop-up window, click **OK**.
4. To bulk delete call rate rules, do as follows:
   a. Select the checkboxes of desired call rate rules, then click **Delete**.
   b. In the pop-up window, click **OK**.

Export and Import Call Rate Rules

The call rate rules configured on Yeastar P-Series Cloud Edition can be exported and saved as a template. You can fill in desired call rate rules in the exported file, and import the file to PBX again. This topic describes how to export and import call rate rules.

Export all the call rate rules

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Under **Rate** tab, click **Export**.

   A CSV file is saved to your computer. To check and edit parameters in the CSV file, see Rate Parameters.

Import call rate rules

We recommend that you export call rate rules to a CSV file first, and use the file as a template to start with. In this way, you can save time and effort.

Prerequisites

Requirements of an imported file:

- **Format**: UTF-8.CSV
- **Size**: Less than 50 MB
- **File name**: Less than 127 characters
- **Import parameters**: Ensure that the imported parameters meet requirements. For more information, see Rate Parameters.

Procedure

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. Under **Rate** tab, click **Import**.
3. In the pop-up window, click **Browse**, and select your CSV file.
4. Click **Import**.
   The call rate rules in the CSV file will be displayed in reverse order in the **Rate** list.

> **Note:**
> The system matches outbound calls with call rate rules from top down. In case of need, you can click ⤒ ^ ⌄ ⤓ to adjust the priority of call rate rules.

Related information

Import and Export -FAQ

Call Accounting Report

Extension Call Accounting Report

Extension Call Accounting report provides a quick overview of the bills for outbound calls made by extensions via specific trunks, and can be broken down to specific extensions, groups of extensions, or organizations of extensions. This topic introduces how to access the report and explains the key metrics in detail.

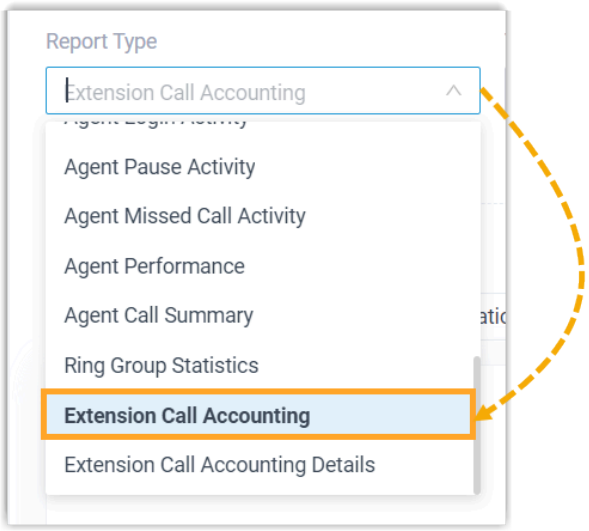
Prerequisites

You have set up at least one call rate rule, otherwise the system wouldn't know how to charge outbound calls.

For more information, see Add a Call Rate Rule.

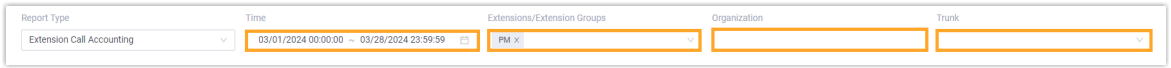
Access Extension Call Accounting report

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. In the **Report Type** drop-down list, select **Extension Call Accounting**.

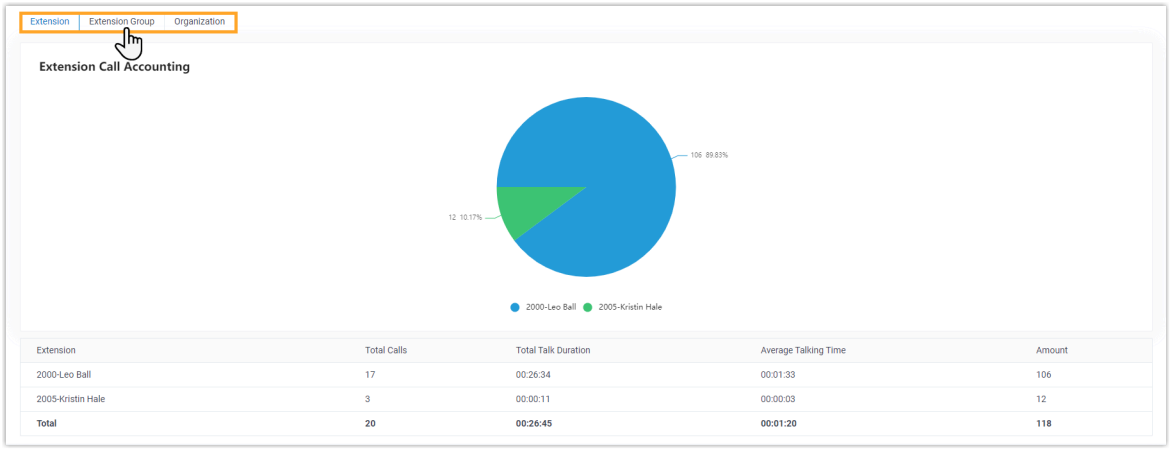


3. Filter data by system time, extension(s), extension group(s), organization(s), or trunk.

> **Note:**
> **Organization** filter is displayed only when you enable **Organization Management** feature (Path: **PBX Settings > Preferences**).



A report that meets the filter criteria is displayed on the page, as shown below.



Note:

- You can click the tab at the top-left corner to switch between the break-down reports.
- If you filter data by extension group(s), the report will NOT include the call statistics from unknown source (the record where **Extension** column is displayed as **Other**).
- If you filter data by organization(s), the report data will include the selected organization and its subordinate departments, but will NOT include the call statistics from unknown source (the record where **Extension** column is displayed as **Other**).

Report details

We take the above report as an example to introduce the key metrics for **Extension Call Accounting** report.

Metric	Description
Total Calls	The total number of outbound calls made where a call rate rule is applied.
Total Talk Duration	The total time between calls answered and calls ended.
Average Talking Time	The average time between calls answered and calls ended.
Amount	The call cost.

Metric	Description
Total (displayed in row)	The sum of all the values in each column. **Note:** • This metric is only available for extension and extension group. • If you select multiple extension groups and one of the extensions belongs to two or more extension groups, the call statistics for the extension will be calculated more than once.

Related information

Extension Call Accounting Details Report

Extension Call Accounting Details Report

Extension Call Accounting Details report provides granular insights into the bills for each outbound call made by extensions via specific trunks. This topic introduces how to access the report and explains the key metrics in detail.

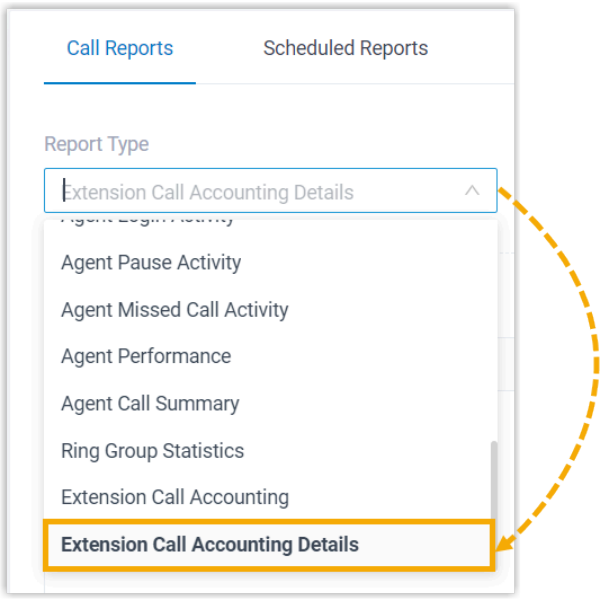
Prerequisites

You have set up at least one call rate rule, otherwise the system wouldn't know how to charge outbound calls.

For more information, see Add a Call Rate Rule.

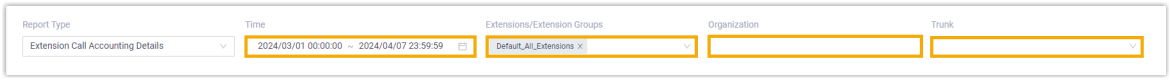
Access Extension Call Accounting Details report

1. Log in to PBX web portal, go to **Reports and Recordings > Call Reports**.
2. In the **Report Type** drop-down list, select **Extension Call Accounting Details**.

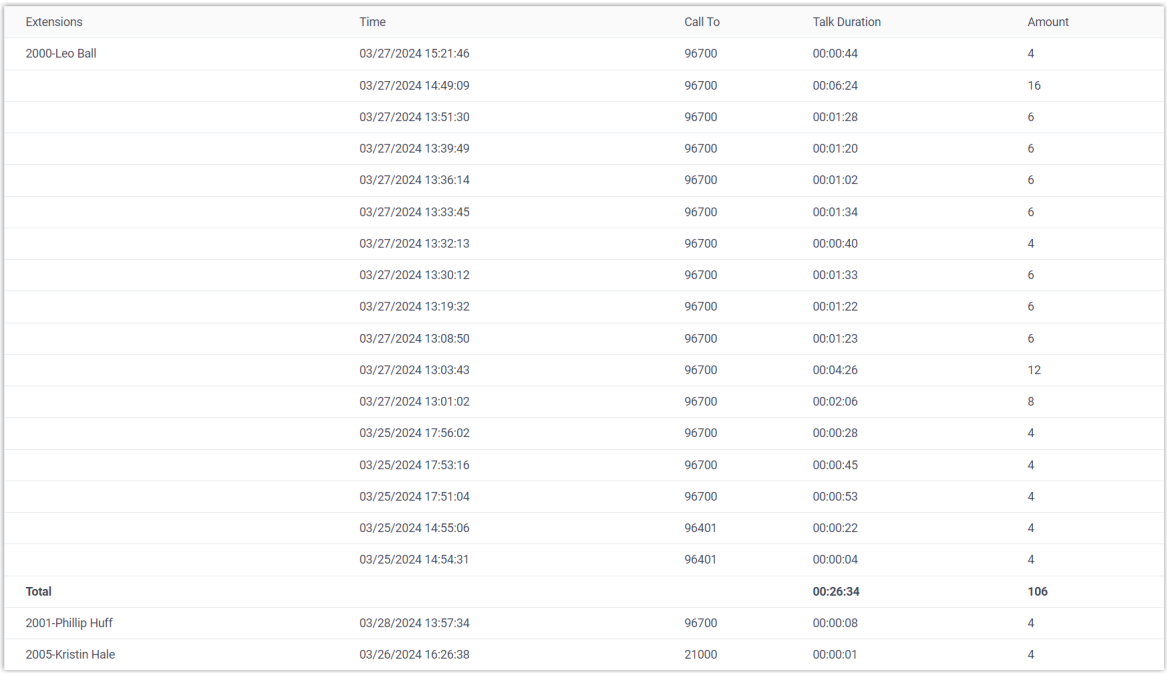


3. Filter data by system time, extension(s), extension group(s), organization(s), or trunk.

> **Note:**
> **Organization** filter is displayed only when you enable **Organization Management** feature (Path: **PBX Settings > Preferences**).



A report that meets the filter criteria is displayed on the page, as shown below.



Extensions	Time	Call To	Talk Duration	Amount
2000-Leo Ball	03/27/2024 15:21:46	96700	00:00:44	4
	03/27/2024 14:49:09	96700	00:06:24	16
	03/27/2024 13:51:30	96700	00:01:28	6
	03/27/2024 13:39:49	96700	00:01:20	6
	03/27/2024 13:36:14	96700	00:01:02	6
	03/27/2024 13:33:45	96700	00:01:34	6
	03/27/2024 13:32:13	96700	00:00:40	4
	03/27/2024 13:30:12	96700	00:01:33	6
	03/27/2024 13:19:32	96700	00:01:22	6
	03/27/2024 13:08:50	96700	00:01:23	6
	03/27/2024 13:03:43	96700	00:04:26	12
	03/27/2024 13:01:02	96700	00:02:06	8
	03/25/2024 17:56:02	96700	00:00:28	4
	03/25/2024 17:53:16	96700	00:00:45	4
	03/25/2024 17:51:04	96700	00:00:53	4
	03/25/2024 14:55:06	96401	00:00:22	4
	03/25/2024 14:54:31	96401	00:00:04	4
Total			**00:26:34**	**106**
2001-Phillip Huff	03/28/2024 13:57:34	96700	00:00:08	4
2005-Kristin Hale	03/26/2024 16:26:38	21000	00:00:01	4

Report details

We take the above report as an example to introduce the key metrics for **Extension Call Accounting Details** report.

Metric	Description
Time	When the outbound call was made.
Call To	The callee number.
Talk Duration	The time between call answered and call ended.
Amount	The call cost.
Total	Total talk duration and amount for extensions that make two or more outbound calls.
Total for All	Total talk duration and amount for the selected extensions. **Note:** This metric is displayed only when you set **Trunk** filter to a specific trunk.

Related information

[Extension Call Accounting Report]

Integration

Speech to Text (STT)

Speech to Text (STT) Overview

Speech to Text, also known as speech recognition, enables transcription of audio messages into texts. Yeastar P-Series Cloud Edition allows you to use a third-party transcription service to implement the audio transcription.

Restriction

This integration is NOT supported on Yeastar P-Series Cloud Edition that can not access Google service.

Supported Service Platform

Yeastar P-Series Cloud Edition supports the following third-party transcription service:

> **Note:**
> The **Speech to Text** feature on Yeastar P-Series Cloud Edition is free. However, you will need to pay for the Speech-to-Text service of the third-party platforms.

- **Google Cloud Speech-to-Text API**

  For more information about the integrations, see Integrate Yeastar P-Series Cloud Edition with Google Cloud Speech-to-Text Service.

Applications

After STT integration is set up on the PBX, the speech recognition can be applied to Voicemail Transcription. Users can receive voicemails in the form of text on different platform:

Linkus UC Clients

Users can check the transcribed text for each voicemail on Linkus Web Client, Linkus Desktop Client, and Linkus Mobile Client.

Email Client

If Voicemail to Email feature is enabled, the transcribed text will be displayed in the email content for received voicemails.

Related information

Enable or Disable Voicemail Transcription

Integrate with Speech to Text (STT) API

Integrate Yeastar P-Series Cloud Edition with Google Cloud Speech-to-Text Service

Before using Voicemail Transcription feature, you need to integrate Yeastar P-Series Cloud Edition with a third-party Speech-to-Text service. This topic describes how to configure the integration of Google Cloud Speech-to-Text (STT) service with Yeastar P-Series Cloud Edition.

Limitations

Region restriction

This integration is NOT supported on Yeastar P-Series Cloud Edition that can not access Google service.

Audio length: 1 minute

The integration of Yeastar P-Series Cloud Edition with Google Cloud Speech-to-Text service uses the Synchronous Recognition method for speech recognition, which can process up to 1 minute of speech audio data.

Service cost

Google Cloud Speech-to-Text service provides a free amount of 60 minutes per month, you will be charged if the minutes of audio processed per month exceeds the free amount. For more information about the pricing, see Google Cloud Speech-to-Text Pricing.

Prerequisites

- You need to create a Google Cloud billing account.
- Make sure the Yeastar P-Series Cloud Edition can access Google services.
  1. Log in to PBX web portal, go to **Maintenance > Troubleshooting > IP Ping**.
  2. In the **Target Host** field, enter `www.google.com`.

3. Click **Start**.
4. Check the **Result** box to see if the packet transmission is normal.
5. Click **Stop** to stop pinging.

Procedure

1. Get the API key from Google Cloud Platform
2. Enable Speech to Text (STT) integration on Yeastar P-Series Cloud Edition

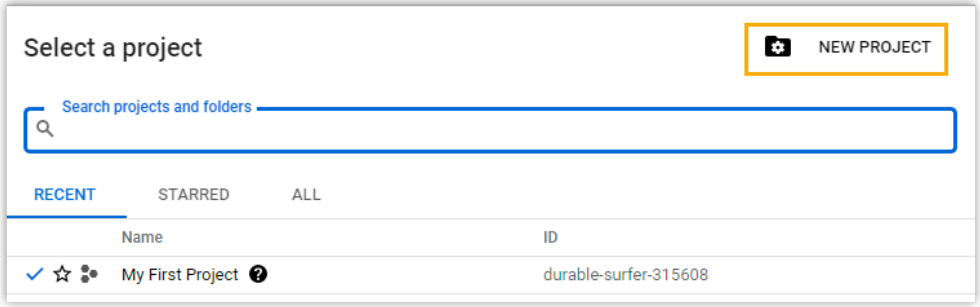
Get the API key from Google Cloud Platform

Step1. Create a project on Google Cloud Platform

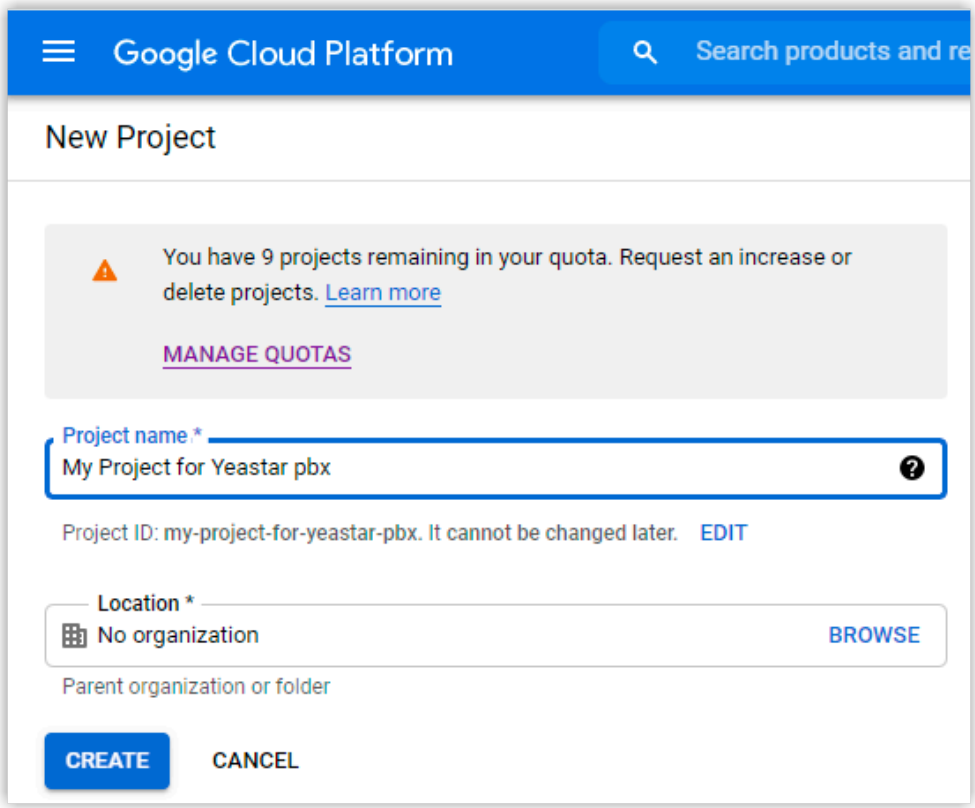
1. Log in to Google Cloud Platform.
2. In the top bar, click **My First Project** to open the project list.



3. On the **Select a project** page, click **NEW PROJECT** in the top-right corner.



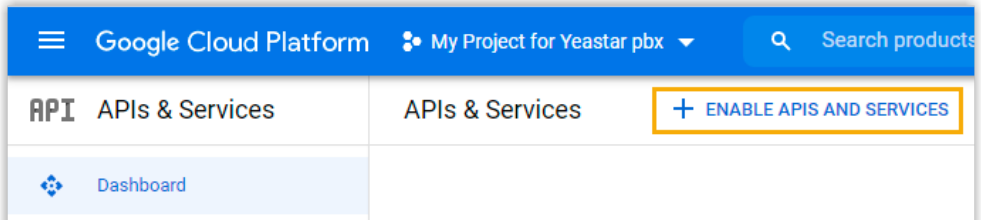
4. On the **New Project** page, set a project name, and click **CREATE**.



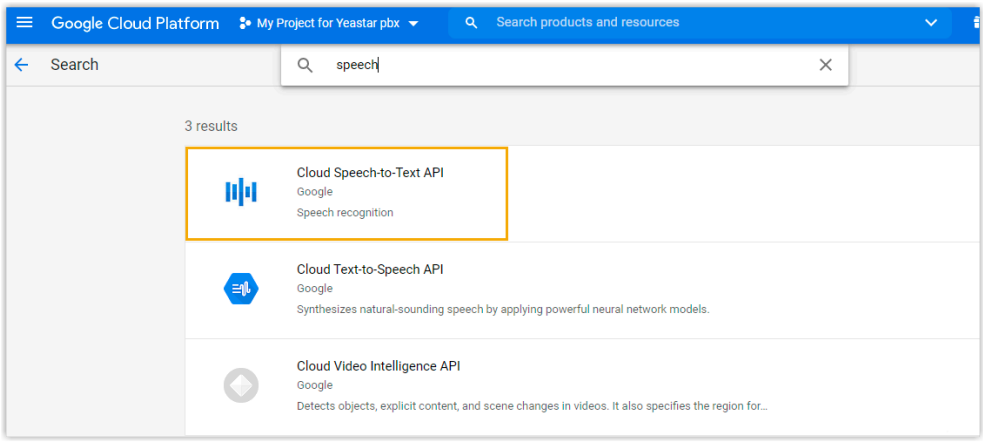
A new project is created, you can select the new project in the project list.

Step2. Enable Speech-to-Text API service on Google Cloud Platform

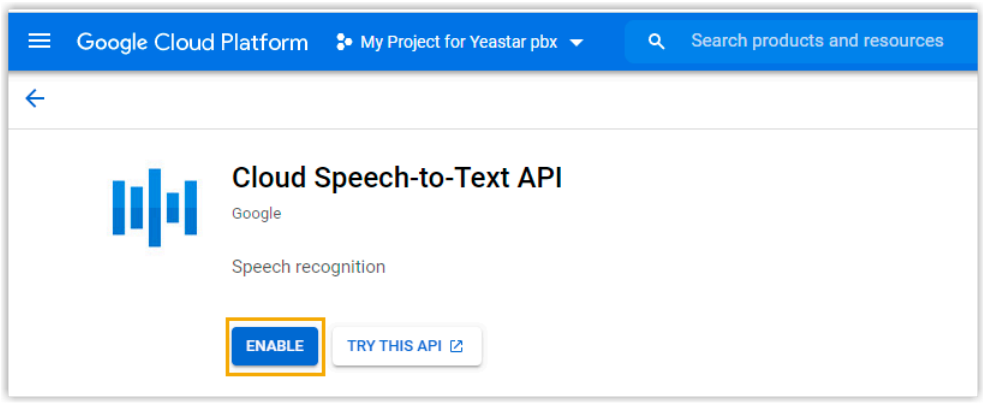
1. In the top-left conner, click ☰ to open the navigation menu, and go to **API & Services > Dashboard**.
2. Click **ENABLE APIS AND SERVICES**.



3. In the API Library, enter speech in the search box and press **Enter**, then select **Cloud Speech-to-Text API**.



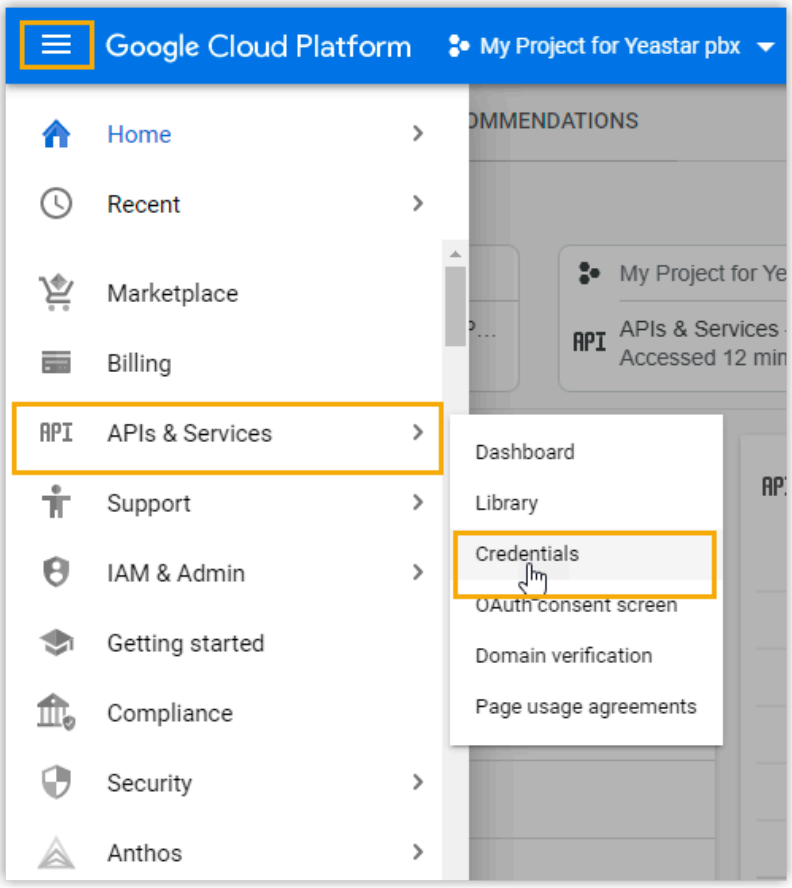
4. Click **ENABLE** button for the **Cloud Speech-to-Text API**.



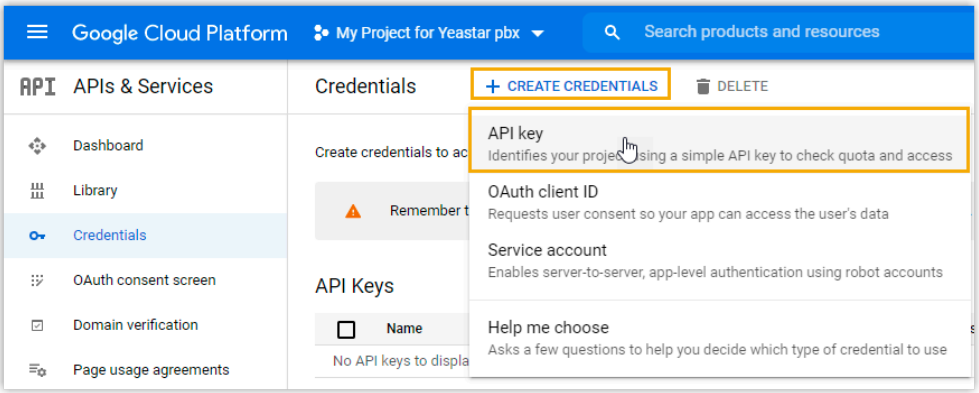
The Speech-to-Text service is enabled.

Step3. Create API credentials on Google Cloud Platform

1. In the left navigation panel, go to **API & Services > Credentials**.

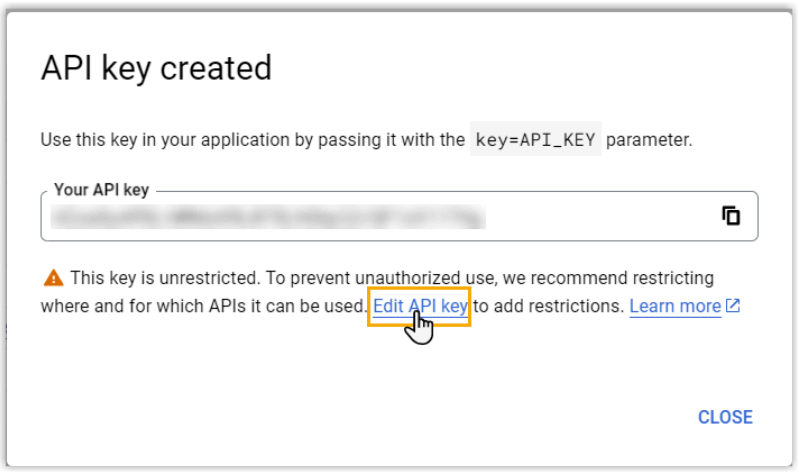


2. Click **CREATE CREDENTIALS** and select **API key**.



3. In the pop-up window, click **Edit API key** to edit and set restrictions for the key.

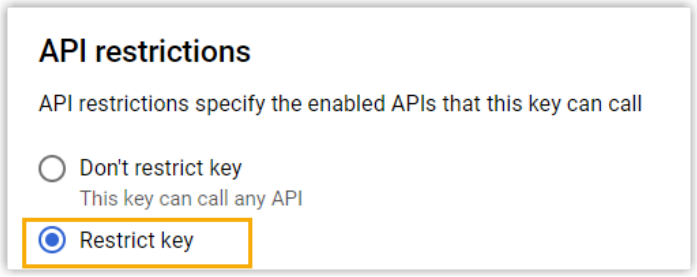
> **Important:**
> For security purpose, you need to restrict your API key, ensuring only authorized requests are made with your API key.



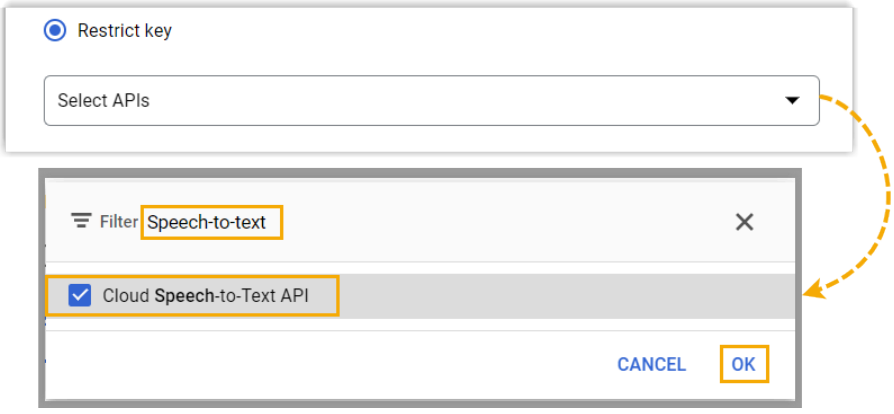
4. In the **Edit API key** page, complete the following configurations.
   a. In the **Name** field, set a name to help you identify the API key.



   b. In the **API restrictions** section, select **Restrict key**.



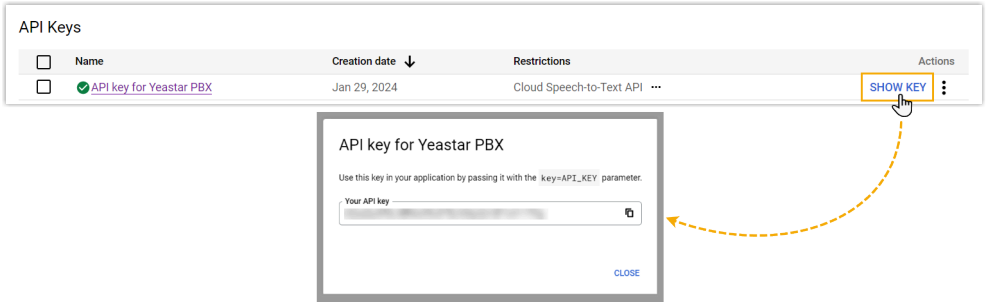
   c. In the **Select APIs** drop-down list, search and select the **Cloud Speech-to-Text API**, then click **OK**.



d. At the bottom of the page, click **SAVE** to apply your configuration.

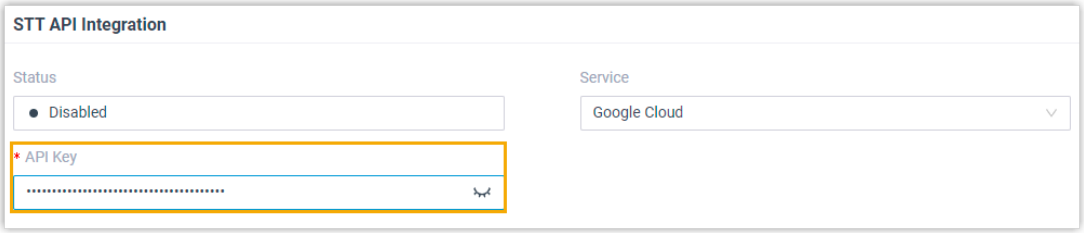
The API key is now only allowed to call the Cloud Speech-to-Text API.

5. In the **Credentials** page, click **SHOW KEY** beside the restricted API key, then in the pop-up window to copy the key.



Enable Speech to Text (STT) integration on Yeastar P-Series Cloud Edition

1. Log in to PBX web portal, go to **Integrations > Speech to Text**.
2. In **STT API Integration** section, fill in the required API credentials.
   - **Service**: Select **Google Cloud**.
   - **API Key**: Paste the restricted API key copied in the former procedure.



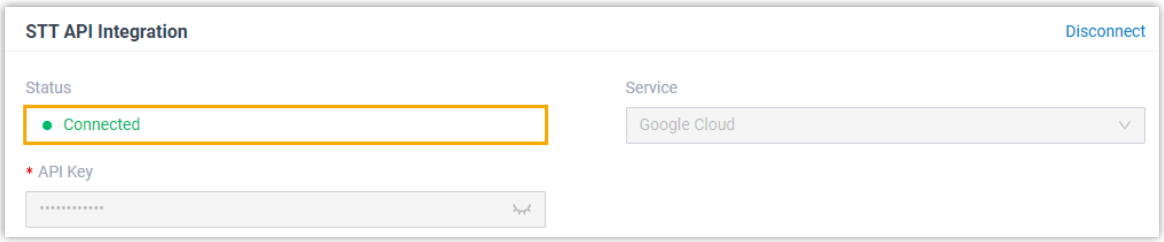
3. In **Settings** section, select the transcription language.

   The audio messages will be transcribed to text in the selected language.

> **Note:**
> If the language of voicemail is different from the selected language, the transcribed text will be inaccurate.

4. Click **Save**.

   If the integration succeeds, the **Status** in the **STT API Integration** section will display **Connected**.



What to do next

After the STT API integration succeeds, go to **Call Features > Voicemail > Voicemail Settings** to enable the Voicemail Transcription feature. For more information, see Enable or Disable Voicemail Transcription.

Related information

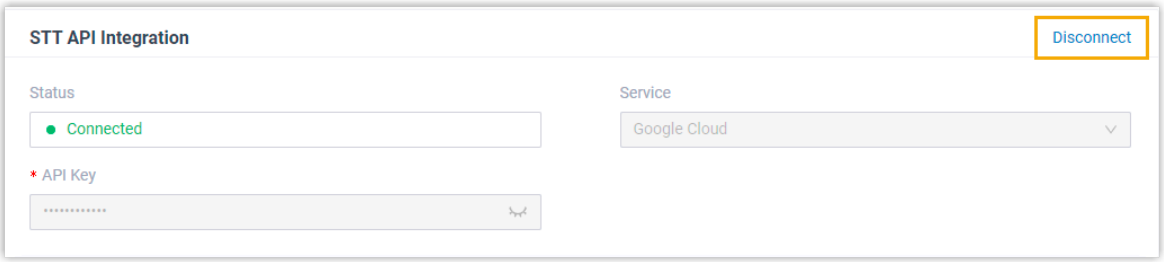
Speech to Text (STT) Overview
Disconnect Speech to Text (STT) API Integration

Disconnect Speech to Text (STT) API Integration

After the STT API integration is connected, you can directly disconnect the API service on PBX if you don't need the Speech to Text feature any more, or want to pause the API service.

Procedure

1. Log in to PBX web portal, go to **Integrations > Speech to Text**.
2. In the **STT API Integration** section, click **Disconnect** in the top-right corner.



3. In the pop-up dialog box, click **Confirm** to disconnect the API service.

   The API integration is disconnected, and the **Status** displays **Disabled**.

Result

The Voicemail Transcription feature is unavailable.

Asterisk Manager Interface (AMI) Overview

The Asterisk Manager Interface (AMI) is a system monitoring and management interface provided by Asterisk. Yeastar P-Series Cloud Edition supports AMI that allows you to connect an AMI client to Yeastar P-Series Cloud Edition.

What is Asterisk Manager Interface (AMI)

Asterisk Manager Interface (AMI) is a standard management interface into Asterisk server. It is a client/server model over TCP that allows a client program to connect to an Asterisk server and issue commands or read events over a TCP/IP stream. With the manager interface, you can control the PBX, originate calls, check mailbox status, monitor extensions and so on.

Connect to Yeastar P-Series Cloud Edition via AMI

1. Enable AMI on PBX.
    a. Log in to PBX web portal, go to **Integrations > AMI**.
    b. Enable **AMI**.
    c. In the **AMI** section, configure the connection authentication.
        - **Username**: Enter the username that can be used by third party to access the AMI of PBX.
        - **Password**: Enter the password that can be used by third party to access the AMI of PBX.
    d. In the **Permitted IP** section, set which clients are allowed to access the AMI of PBX.
        i. In the **IP Address** field, click **Add**.
        ii. Enter the IP address or IP section that is allowed to access the AMI of PBX.

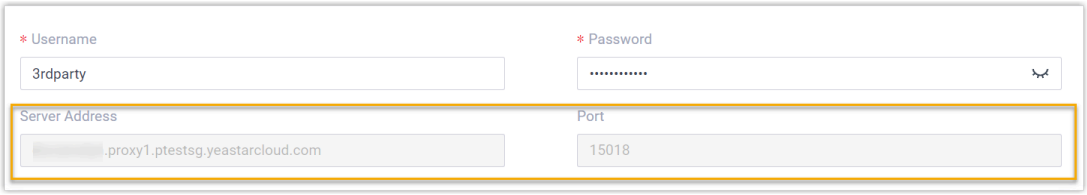
        The input format should be *XXX.XXX.XXX.XXX*.

        For example: IP address *216.207.245.47* with subnet mask *255.255.255.255* means that only the device with IP address 216.207.245.47 is allowed to access the PBX via AMI.

        > **Note:**
        > You can add up to 4 permitted IP addresses.

    e. Click **Save** and **Apply**.

The **Server Address** and **Port** of the PBX will be displayed on the web page and can NOT be modified.



2. Configure AMI client with the authentication information provided on PBX, and connect client to PBX.

Database Grant

Database Grant Overview

Yeastar P-Series Cloud Edition is based on MySQL database. Database Grant is a feature that allows you to grant permissions for a third-party software to access the PBX database.

Applications

Database Grant is usually applied in the following scenarios:

- **Billing System**

  By accessing the PBX database, you can get CDR and save it to the local database of billing software. Then you can charge calls by CDR.
- **Call Center**

  Get CDR and save it to the local database of call center software.

Limitation

After accessing the PBX database, only cdr data is available to be checked and downloaded, other data cannot be accessed.

Get CDR Data from Database of Yeastar P-Series Cloud Edition

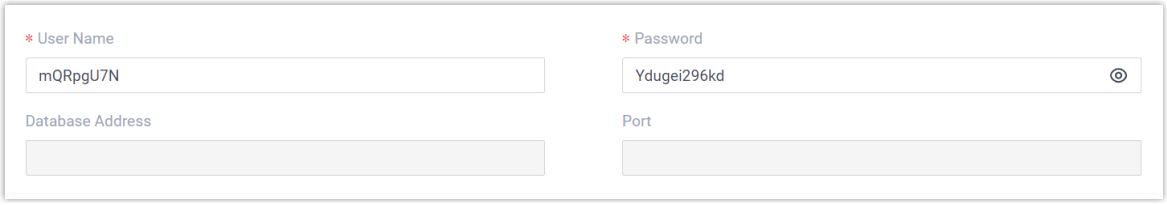
Yeastar P-Series Cloud Edition allows you to access the system database and get CDR data. This topic describes how to get CDR data from the PBX database via Navicat software.

Procedure

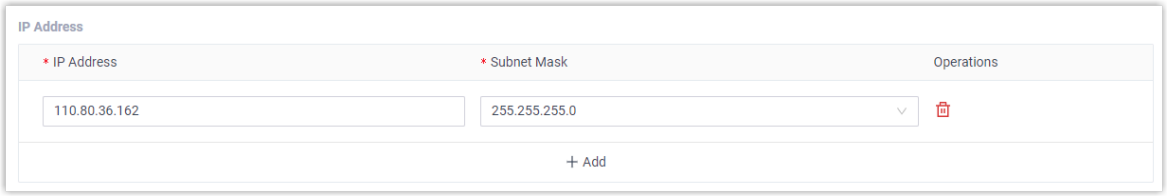
1. Grant access to the PBX database
2. Access the PBX database via Navicat software

Grant access to the PBX database

1. Log in to PBX web portal, go to **Integration > Database Grant**.
2. Turn on **Database Grant** option and configure the authentication information for the third-party software to access the PBX database.



   - **User Name**: Use the randomly generated user name or change the name.
   - **Password**: Use the randomly generated password or change password.
3. In the **Permitted IP** section, configure which IP addresses are allowed to access the database.



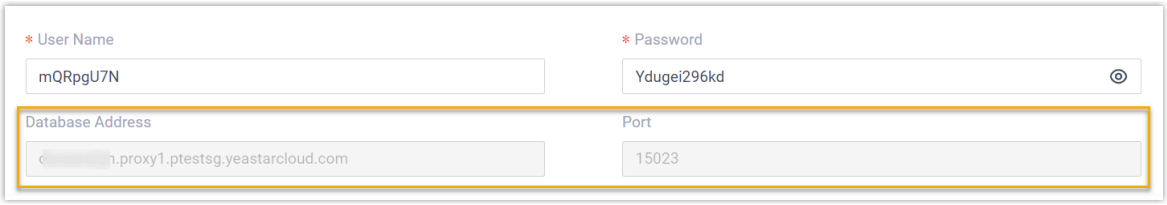
   a. Click **Add**.
   b. Enter the permitted IP address and subnet mask.

      In this example, enter the public IP address of the current network `110.80.36.162` and subnet mask `255.255.255.0` to allow all IP addresses in thecurrent network to access the database.

> **Note:**
> Restricted from MySQL database, only the two subnet masks are allowed to be filled in: `255.255.255.255` and `255.255.255.0`.

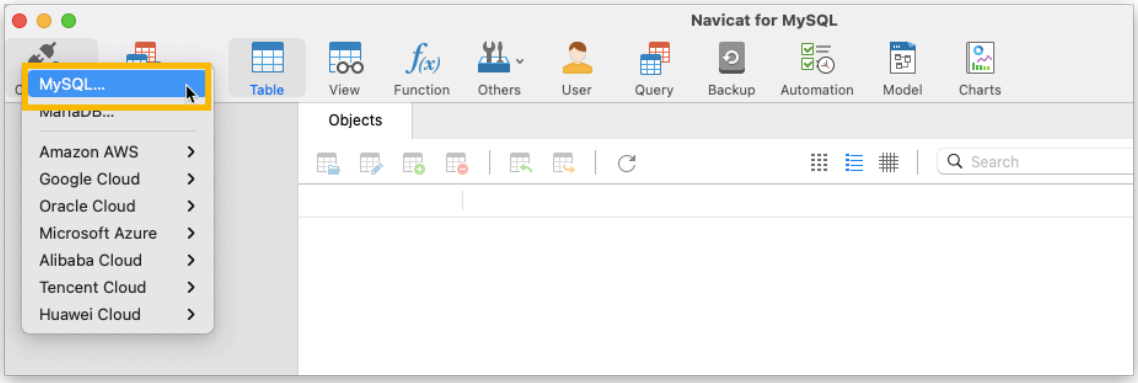
4. Click **Save** and **Apply**.

The **Database Address** and **Port** of the PBX will be displayed on the web page and can not be modified.

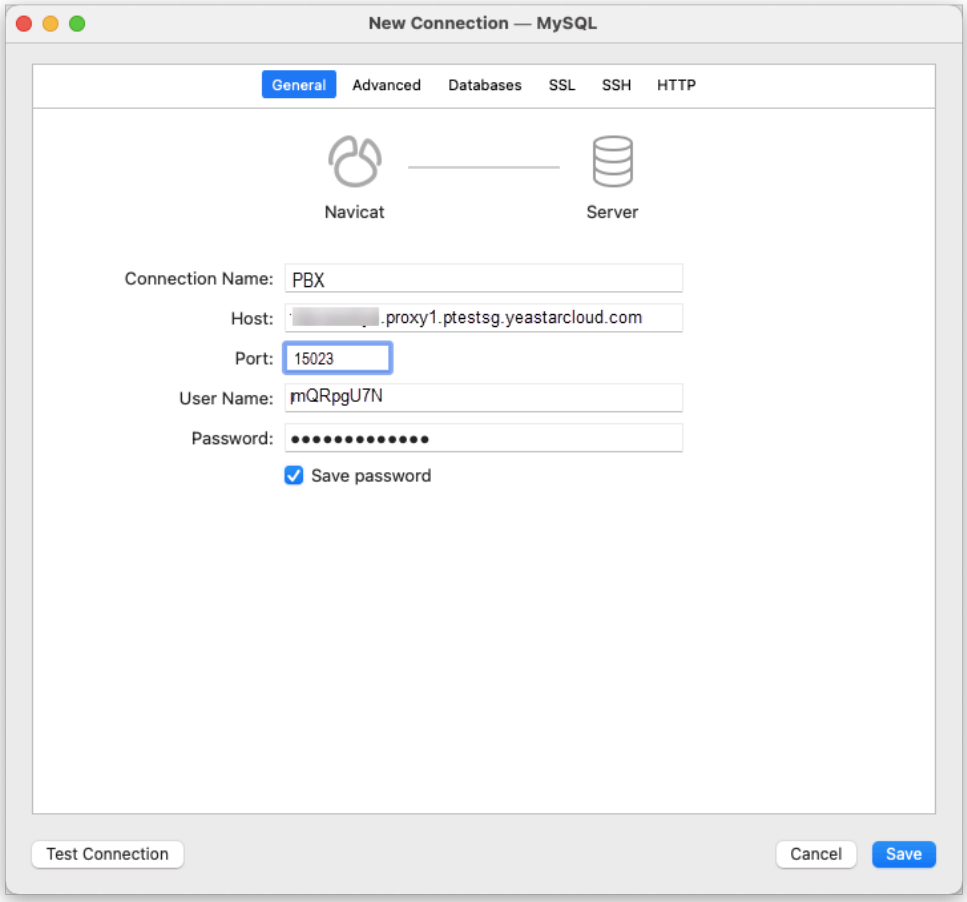


Access the PBX database via Navicat software

1. Launch Navicat for MySQL on the PC.
2. On the Navicat for MySQL, click **Connection** and select **MySQL**.



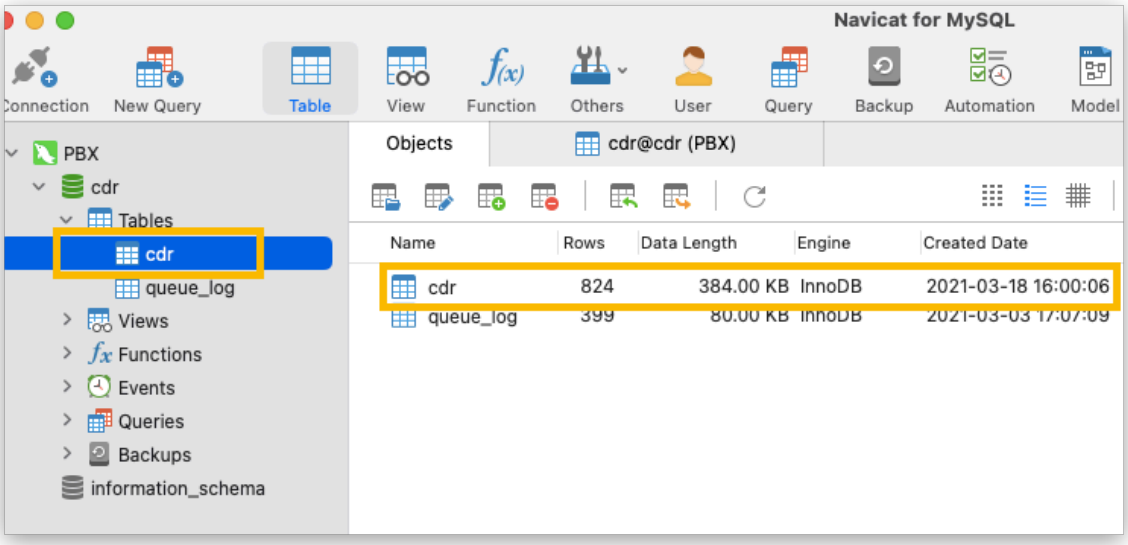
3. In the pop-up window, enter the following information:



- **Connection Name**: Enter a connection name to help you identify it.
- **Host**: Enter the database address of PBX
- **Port**: Enter `15023`.
- **User Name**: Enter the user name that is configured on the PBX. In this example, enter `mQRpgU7N`.
- **Password**: Enter the password that is configured on the PBX. In this example, enter `Ydugei296kd`.

4. Click **Save**.
5. To check CDR data, double click the new connection, and select cdr table.

   For more information about the cdr table, see cdr Table in the PBX Database.



cdr Table in the PBX Database

This topic describes details of cdr table stored in the database of Yeastar P-Series Cloud Edition.

Field	Descriptions
id	System internal flag
datetime	Date and time
timestamp	System internal flag
uid	System internal flag
clid	System internal flag
src	Caller's number
srcname	Caller's name
srcaddr	System internal flag
dst	Callee's number
dstname	Callee's name
dcontext	System internal flag
channel	System internal flag
dstchannel	System internal flag
srctrunk	Source trunk

Field	Descriptions
dsttrunk	Destination trunk
lastapp	System internal flag
lastdata	System internal flag
duration	Total duration of the call (calculates from the beginning of the call)
ringduration	Ringing duration of the call
talkduration	Talk duration of the call (calculates after the call is answered)
disposition	Call status: • NO ANSWER • FAILED • BUSY • ANSWERED • VOICEMAIL • CONGESTION
amaflags	System internal flag
calltype	Communication Type • Internal • Inbound • Outbound • Callback
accountcode	Pin code
uniqueid	System internal flag
didnumber	DID number
dodnumber	DOD number
recordfile	Recording file name
recordpath	Recordings path (with file name)
srcchanurl	Caller's SIP URI
dstchanurl	Callee's SIP URI
reasonpartya	System internal flag
reasonpartyb	System internal flag
reasonpartyc	System internal flag
reasonpartyd	System internal flag
reasonpartye	System internal flag

Field	Descriptions
reasonpartyf	System internal flag
displayonweb	System internal flag
src_del_cdr	System internal flag
dst_del_cdr	System internal flag
src_del_recording	System internal flag
dst_del_recording	System internal flag
srcnameprefix	System internal flag
dstnameprefix	System internal flag
misscall_isread	System internal flag
in2outbound	System internal flag
concurrentcalls	System internal flag
videocall	System internal flag
rascall	System internal flag
tryvideocall	System internal flag

References

Import and Export Parameters Overview

Check the required parameters, optional parameters, and restrictions in the import and export files.

Background information

CSV (comma-separated values) files can expedite the bulk creation of various settings. A CSV file is a plain text file that stores tabular data from database-style tools, such as Excel.

Yeastar P-Series Cloud Edition allows you to export data as a CSV file, specify data in the CSV file, and import the file to PBX to modify settings in bulk, such as creating extensions in bulk using CSV file.

Which features support importing and exporting data

Yeastar P-Series Cloud Edition supports importing and exporting data of the following modules:

> **Note:**
> The supported parameters are different depending on firmware version.

- Extension Parameters
- Organization Parameters
- Contacts Parameters
- Holidays Parameters
- Speed Dial Number Parameters
- Emergency Number Parameters
- Auto Provisioning Phone Information Parameters
- Trunk Parameters
- Trunk DIDs/DDIs Parameters
- Trunk Outbound Caller ID Parameters
- Inbound Caller ID Reformatting Rule Parameters
- Inbound Route Parameters
- DID Number to Specific Extension Parameters
- Outbound Route Parameters
- Outbound Call Frequency Restriction Rule Parameters

- Rate Parameters
- Allowed Numbers Parameters
- Blocked Numbers Parameters

Extension Parameters

Descriptions for parameters in exported and imported Extension CSV file.

Parameter	Description	Importance	Restriction	Default Value
First Name	The first name of extension user.	At lease one is required	The maximum character length is 63. **Note:** **First Name** will be filled with a value of *Extension Number* if you leave these fields empty.	Extension number
Last name	The last name of extension user.			N/A
Email Address	The email address of extension user.	Optional	Only numbers, letters, and characters `@ _ - . '` are allowed. Must start with a number, letter, or character _ and follow the email address format XXX@XXX.XX. Extension's email address cannot be duplicated. The maximum character length is 255.	N/A
Mobile Number	The mobile number of extension user.	Optional	Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.	N/A
User Password	The password for extension user to log in to Linkus client and PBX web portal.	Required	Must contain numbers, uppercase, and lowercase letters. **Restriction:** Do not allow four consecutive digits, four same digits, extension number, first name, or last name. The minimum character length is 10 and the maximum is 63.	Generate Randomly

Parameter	Description	Importance	Restriction	Default Value
			Note: **User Password** will be generated randomly if you leave this field empty.	
Organization	The organization to which the extension user belongs.	Optional	Permitted value: The organization names that existed in PBX. **Note:** • **Organization** will be filled with default value if you leave this field empty. • When entering the organization name, it must be the full path of parent organization, connected by `/`. For multiple organizations, please use `&` to separate. Examples are given below: • If belong to root organization "Yeastar", enter `Yeastar`. • If belong to first-layer organization "Marketing Center", enter `Yeastar/Marketing Center`. • If belong to second-layer organization "Support Team", enter `Yeastar/Marketing Center/Support Team`. • If belong to multiple organizations, enter `Yeastar/Marketing Center&Yeastar/Product Management Center`.	Root organization, namely the Company Name.

Parameter	Description	Importance	Restriction	Default Value
User Role	The role for extension user with PBX management permission.	Required	Permitted value: `0` or one of the role names defined in the PBX. `0` means [None]. **Note:** **User Role** will be filled with default value `0` if you leave this field empty.	0
Extension Number	The extension's number.	Required	Extension Number cannot be duplicated, and only numbers are allowed. The maximum character length is 7.	N/A
Caller ID	The caller ID that is displayed on the callee's device.	Required	Numbers, letters, and special characters `( ) . - + * #` are allowed. The maximum character length is 31. **Note:** **Caller ID** will be filled with default value `Extension Number` if you leave this field empty.	Extension Number
Registration Name	The registration name that is used to validate extension registration.	Required	The maximum character length is 63. **Note:** **Registration Name** will be generated randomly if you leave this field empty.	Generate Randomly
Registration Password	The password for the user to register the SIP extension.	Required	Must contain numbers, uppercase, and lowercase letters. **Restriction:** Do not allow four consecutive digits, four same digits, extension number, first name, or last name. The minimum character length is 10 and the maximum is 63.	Generate Randomly

Parameter	Description	Importance	Restriction	Default Value
			Note: **Registration Password** will be generated randomly if you leave this field empty.	
IP Phone Concurrent Registrations	How many SIP phones are allowed to register with the extension.	Required	Permitted value: • *1*: Allow one phone to register with the extension. • *2*: Allow two phones to register with the extension. • *3*: Allow three phones to register with the extension. **Note:** **IP Phone Concurrent Registrations** will be filled with default value *1* if you leave this field empty.	1
Emergency Outbound Caller ID	The outbound Caller ID for the extension when it makes emergency calls.	Optional	Numbers, letters, and characters *( ) . - + * #* are allowed. The maximum character length is 31.	N/A
Enable Voicemail	Whether to enable or disable voicemail feature.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Enable Voicemail** will be filled with default value *1* if you leave this field empty.	1
Voicemail PIN Authentication	Whether to enable or disable voicemail PIN authentication.	Required if **Enable Voicemail** = *1*	Permitted value: • *0*: Disable • *1*: Enable **Note:**	1

Parameter	Description	Importance	Restriction	Default Value
			Voicemail PIN Authentication will be filled with default value *1* if you leave this field empty.	
Voicemail Access PIN	The PIN for authentication when accessing voicemail box.	Required if **Enable Voicemail** = *1* & **Voicemail PIN Authentication** = *1*	Only numbers are allowed. The minimum character length is 3 and the maximum is 15. **Note:** **Voicemail Access PIN** will be generated randomly if you leave this field empty.	Generate Randomly
Voicemail Language	The system prompt language used by the extension's voicemail.	Optional	Permitted value: `follow_system` or one of the prompt language existed in PBX System Prompt (e.g. `German`). **Note:** **Voicemail Language** will be filled with default value *follow_system* if the language you enter does not exist in PBX, or if you leave this field empty.	follow_system
New Voicemail Notification	The notification type for new voicemail.	Required if **Enable Voicemail** = *1*	Permitted value: • *no*: No Email Notifications • *with_attach*: Send Email Notifications with Attachment • *without_attach*: Send Email Notifications without Attachment **Note:** **New Voicemail Notification** will be filled with default value *no* if you leave this field or Email Address empty.	no

Parameter	Description	Importance	Restriction	Default Value
Send to	Specify the email address for receiving notification emails about new voicemails.	Required if **Enable Voicemail** = *1* & **New Voicemail Notification** = *with_attach* or *without_attach*	Permitted value: • *user Email*: Send notification emails to the user's email address. • *custom Email*: Send notification emails to a custom email address.	user Email
Voicemail Email Address	Enter a custom email address for receiving notification emails about new voicemails.	Required if **Enable Voicemail** = *1* & **New Voicemail Notification** = *with_attach* or *without_attach* & **Send to** = *custom Email*	Only numbers, letters, and characters *@ _ - .* are allowed. Must start with a number, letter, or character _ and follow the email address format XXX@XXX.XX. The maximum character length is 255.	N/A
After Notification	The way to handle voicemail message in mailbox after receiving the message notification via email.	Required if **Enable Voicemail** = *1* & **New Voicemail Notification** = *with_attach*	Permitted value: • *no*: Do Nothing • *mark_read*: Mark as read • *delete*: Delete Voicemail **Note:** **After Notification** will be filled with default value *no* if you leave these fields empty.	no
		Required if **Enable Voicemail** = *1* & **New Voicemail**	Permitted value: • *no*: Do Nothing • *mark_read*: Mark as read	no

Parameter	Description	Importance	Restriction	Default Value
		Notification = *without_attach*	**Note:** **After Notification** will be filled with default value *no* if you leave these fields empty.	
Play Date and Time	Whether to announce arrival time of the message before playing the voicemail message.	Required if **Enable Voicemail** = *1*	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Play Date and Time** will be filled with default value *0* if you leave this field empty.	0
Play Caller ID	Whether to announce caller ID of the party that left the message before playing the voicemail message.	Required if **Enable Voicemail** = *1*	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Play Caller ID** will be filled with default value *0* if you leave this field empty.	0
Play Message Duration	The duration of the message (in minutes) will be announced before playing the voicemail message.	Required if **Enable Voicemail** = *1*	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Play Message Duration** will be filled with default value *0* if you leave this field empty.	0
Send email notification when the User	Whether to send email notification when the User	Required	Permitted value: • *0*: Disable • *1*: Enable	1

Parameter	Description	Importance	Restriction	Default Value
Password is changed	Password is changed.		**Note:** **Send email notification when the User Password is changed** will be filled with default value `1` if you leave this field empty.	
Send email notifications on missed calls	Whether to send email notifications on missed calls.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Send email notifications on missed calls** will be filled with default value `0` if you leave this field empty.	0
Recording operation	Whether to allow users to switch their own recording status during a call.	Required	Permitted value: • `0`: No permission • `1`: Pause/Resume • `2`: Start/Pause/Resume **Note:** **Recording operation** will be filled with default value `0` if you leave this field empty.	0
All Busy Mode for Endpoints	Whether to forward a new incoming call to the Busy destination when one of the endpoints with extension registered is busy in a call.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **All Busy Mode for Endpoints** will be filled with default value `0` if you leave this field empty.	0

Parameter	Description	Importance	Restriction	Default Value
All Reject Mode for Endpoints	Whether to stop the incoming call from ringing other endpoints and forward to the Busy destination when it was rejected on an endpoint.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **All Reject Mode for Endpoints** will be filled with default value `0` if you leave this field empty.	0
Call Popup URL	Whether to automatically open a custom URL (web page) upon receiving an incoming call.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Call Popup URL** will be filled with default value `0` if you leave this field empty.	0
Popup URL	The address of third-party URL, followed by the variables that you want to pass.	Required if **Call Popup URL** = `1`	The maximum character length is 255.	http://example.com/somepage.php?number={{.CallerNumber}}&name={{.CallerDisplayName}}
Communication type	The types of calls that will trigger the call popup.	Required if **Call Popup URL** = `1`	**Permitted value**: `Internal` and `Inbound`. • For multiple types, enter values in order and use `&` as a separator, e.g. Internal & Inbound. • If the value you enter is not permitted, it will be skipped.	Inbound
Trigger Event	When the call popup will be automatically triggered.	Required if **Call Popup URL** = `1`	**Permitted value**: `Ringing`, `Answered`, and `Call End`. **Note:**	Ringing

Parameter	Description	Importance	Restriction	Default Value
			Trigger Event will be filled with default value `Ringing` if you leave the field empty.	
Allow Being Monitored	Whether to allow the user's calls to be monitored.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Allow Being Monitored** will be filled with default value `1` if you leave this field empty.	1
Video Preview	Whether to allow other extension users to click a specific button to preview the extension's video when receiving a call from the extension.	Optional	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Video Preview** will be filled with default value `0` if you leave this field empty.	0
Auto Preview	Whether to allow other extension users to automatically preview the extension's video when receiving a call from the extension.	Optional	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Auto Preview** will be filled with default value `0` if you leave this field empty.	0
Hot Desking	Whether to allow the extension to log in to a hot desking phone.	Optional	Permitted value: • `0`: Disable • `1`: Enable **Note:**	0

Parameter	Description	Importance	Restriction	Default Value
			Hot Desking will be filled with default value *0* if you leave this field empty.	
Log out of Queue	Whether to automatically log out the dynamic agent from the queue when the user log out of a hot desking phone.	Optional	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Log out of Queue** will be filled with default value *0* if you leave this field empty.	0
Automatic Guest Out	Whether to automatically log out the extension from a hot desking phone.	Optional	Permitted value: • `0`: Never • `1`: After • `2`: At Daily **Note:** **Automatic Guest Out** will be filled with default value `0` if you leave this field empty.	0
After hr.	The hour of the specified time period for automatically logging out the extension from a hot desking phone.	Required if **Automatic Guest Out** = *1*	Permitted value: 00 - 23 **Note:** **After hr.** will be filled with default value `08` if you leave this field empty.	08
After min.	The minute of the specified time period for automatically logging out the extension from a hot desking phone.	Required if **Automatic Guest Out** = *1*	Permitted value: 00 - 59 **Note:** **After min.** will be filled with default value `00` if you leave this field empty.	00

Parameter	Description	Importance	Restriction	Default Value
At Daily	The time for automatically logging out the extension from a hot desking phone on a daily basis.	Required if **Automatic Guest Out** = *2*	Time Format: HH:MM (e.g. 13:30) **Note:** The End Time must be later than the Start Time.	N/A
DTMF Mode	The mode for sending DTMF tones.	Required	Permitted value: *rfc4733*, *info*, *inband* or *auto*. **Note:** **DTMF Mode** will be filled with default value *rfc4733* if you leave this field empty.	rfc4733
Transport	The protocol for transport.	Required	Permitted value: *udp*, *tcp*, or *tls*. **Note:** **Transport** will be filled with default value *udp* if you leave these fields empty.	udp
Qualify	Whether to send the SIP OPTIONS packet periodically to the SIP device to check if the device is online.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Qualify** will be filled with default value *1* if you leave this field empty.	1
T.38 Support	Whether to support T.38 fax for this extension.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **T.38 Support** will be filled with default value `0` if you leave this field empty.	0

Parameter	Description	Importance	Restriction	Default Value
SRTP	Whether to encrypt RTP packets.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **SRTP** will be filled with default value `0` if you leave this field empty.	0
Disable Outbound Calls	Whether to restrict the user from making outbound calls.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Disable Outbound Calls** will be filled with default value `0` if you leave this field empty.	0
Disable Outbound Calls outside Business Hours	Whether to restrict the user from making outbound calls outside business hours.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Disable Outbound Calls outside Business Hours** will be filled with default value `0` if you leave this field empty.	0
Disallow International Calls	Whether to restrict the user from making international calls.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Disable International Calls** will be filled with	1

Parameter	Description	Importance	Restriction	Default Value
			default value `1` if you leave this field empty.	
Outbound Route Permission	Specify the outbound routes that this extension is allowed to use.	Optional	Permitted value: one or more outbound route names existed in PBX. **Note:** • If the outbound route name you enter does not exist in PBX, it will be skipped. • For multiple outbound routes, please enter outbound route names and use `&` as a separator, e.g. name1&name2.	N/A
Max Outbound Call Duration (s)	The maximum call duration in seconds for making outbound calls from this extension.	Required	Only numbers are allowed. Specially, `-1` means follow system and `0` means unlimited. The maximum character length is 7. **Note:** **Max Outbound Call Duration (s)** will be filled with default value `-1` if you leave these fields empty.	-1
Outbound Call Frequency Restriction	The restriction rule(s) that used to limit the extension outbound call frequency within specified time period.	Optional	Permitted value: One or more Outbound Call Frequency Restriction names existed in PBX. **Note:** • Use `&` to separate multiple names, e.g. name1&name2.	Default_Ext_Outbound Call Frequency

Parameter	Description	Importance	Restriction	Default Value
			• If you leave this field empty, it will be filled with default value. • If the names you entered are not existing in PBX, it will be skipped.	
Linkus Mobile Client	Whether to allow the extension user to log in to Linkus Mobile Client.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Linkus Mobile Client** will be filled with default value *1* if you leave this field empty.	1
Linkus Desktop Client	Whether to allow the extension user to log in to Linkus Desktop Client.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Linkus Desktop Client** will be filled with default value *1* if you leave this field empty.	1
Linkus Web Client	Whether to allow the extension user to log in to Linkus Web Client.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Linkus Web Client** will be filled with default value *1* if you leave this field empty.	1
Linkus Pad Client (SDK)	Whether to allow the extension user	Required	Permitted value: • *0*: Disable	1

Parameter	Description	Importance	Restriction	Default Value
	to log in to Linkus Pad Client (SDK).		• *1*: Enable **Note:** • This parameter only takes effect when Linkus SDK is enabled (Path: **Integrations > Linkus SDK**). • **Linkus Pad Client (SDK)** will be filled with default value *1* if you leave this field empty.	
Linkus Mobile Client Codec	Specify the audio codec for this user's Linkus Mobile Client.	Optional	Permitted value: • *ulaw* • *alaw* • *ilbc* • *g722* • *g729*	N/A

Related information

Export and Import SIP Extensions
Import and Export -FAQ

Organization Parameters

Descriptions for parameters in exported and imported Organization CSV file.

Parameter	Importance	Restriction	Default Value
Organization Name	Required	The maximum character length is 127.	N/A
Parent Organization	Required	Permitted value: The full path of parent organization. For multiple organizations, please use & to separate.	N/A

Parameter	Importance	Restriction	Default Value
		Examples are shown as below: • If belong to root organization "Yeastar", enter `Yeastar`. • If belong to first-layer organization "Marketing Center", enter `Yeastar/Marketing Center`. • If belong to second-layer organization "Support Team", enter `Yeastar/Marketing Center/Support Team`. • If belong to multiple organizations, enter `Yeastar/Marketing Center&Yeastar/Product Management Center`.	

Related information

Export and Import Organizations
Import and Export -FAQ

Contacts Parameters

Descriptions for parameters in exported and imported Company Contacts CSV file and Personal Contacts CSV file.

Parameter	Importance	Restriction
First Name	At lease one is required	The maximum character length is 127 (63 for first name and 63 for last name).
Last Name		
Company Name	Optional	The maximum character length is 127.
Email	Optional	Only numbers, letters, and characters `@ _ - . '` are allowed. Must start with a number, letter, or character _ and follow the email address format XXX@XXX.XX. The maximum character length is 255.
Business Number	At lease one is required	Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Business Number 2		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Business Fax		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.

Parameter	Importance	Restriction
Mobile		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Mobile 2		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Home		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Home 2		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Home Fax		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
Other		Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.
ZIP Code	Optional	The maximum character length is 255.
Street	Optional	The maximum character length is 255.
City	Optional	The maximum character length is 255.
State	Optional	The maximum character length is 255.
Country	Optional	The maximum character length is 255.
Remark	Optional	The maximum character length is 1024.
Phonebook	Optional	**Permitted value**: One or more phonebook names existed in PBX. For multiple phonebooks, enter the names and use `&` as a separator, e.g. phonebook_name1&phonebook_name2. **Note:** • **Phonebook** will be filled with default value `Default_All_Contacts` if you leave these fields empty. • System will create new phonebook(s) if you fill in a name that doesn't exist.

Related information

Export and Import Company Contacts
Linkus Web Client Guide - Export personal contacts
Linkus Web Client Guide - Import personal contacts

Import and Export -FAQ

Holidays Parameters

Descriptions for parameters in exported and imported Holidays CSV file.

Parameter	Importance	Restriction	Default Value
Name	Required	The maximum character length is 127. The holiday name can not be duplicated.	N/A
Holiday Type	Required	Permitted value: • *By Date* • *By Month* • *By Week* **Note:** **Holiday Type** will be filled with default value *By Date* if you leave this field empty.	By Date
Start Date (By Date)	Required if **Holiday Type** = *By Date*	Format: DD/MM/YYYY HH:MM E.g. 25/10/2020 13:30	N/A
End Date (By Date)		**Note:** The End Date must be later than the Start Date.	N/A
Start Date (By Month)	Required if **Holiday Type** = *By Month*	Format: DD/MM HH:MM E.g. 31/10 13:30	N/A
End Date (By Month)		**Note:** The End Date must be later than the Start Date.	N/A
Month (By Week)	Required if **Holiday Type** = *By Week*	Permitted value: • *January* • *February* • *March* • *April* • *May*	N/A

Parameter	Importance	Restriction	Default Value
		• *June* • *July* • *August* • *September* • *October* • *November* • *December*	
Week (By Week)		Permitted value: • *First Week* • *Second Week* • *Third Week* • *Fourth Week* • *Last Week*	N/A
Day (By Week)		Permitted value: • *Sunday* • *Monday* • *Tuesday* • *Wednesday* • *Thursday* • *Friday* • *Saturday*	N/A
Start Time (By Week)		Format: HH:MM E.g. 13:30	N/A
End Time (By Week)		**Note:** The End Time must be later than the Start Time.	N/A

Related information

Export and Import Holidays

Speed Dial Number Parameters

Descriptions for parameters in exported and imported Speed Dial Number CSV file.

Parameter	Importance	Restriction
Speed Dial Number	Required	The maximum character length is 4.

Parameter	Importance	Restriction
		Only numbers and characters * # are allowed. Speed dial number cannot be duplicated.
Phone Number	Required	The maximum character length is 31. Numbers, letters, and characters ( ) . - + * # are allowed.

Related information

Export and Import Speed Dial Numbers
Import and Export -FAQ

Emergency Number Parameters

Descriptions for parameters in exported and imported Emergency Number CSV file.

Parameter	Importance	Restriction	Default Value
Name	Required	The maximum character length is 63. Characters ; " , \ are not allowed. Emergency number's name cannot be duplicated.	N/A
Emergency Number	Required	The maximum character length is 31. Numbers, letters, and characters ( ) . - + * # are allowed. Emergency number cannot be duplicated.	N/A
Emergency Outbound Caller ID Priority	Required	Permitted value: • *emergency_first*: Trunk's Emergency Outbound Caller ID • *ext_first*: Extension's Emergency Outbound Caller ID **Note:** **Emergency Outbound Caller ID Priority** will be filled with default value *emergency_first* if you leave this field empty.	emergency_first

Parameter	Importance	Restriction	Default Value
Trunk	Required	Permitted value: one of trunks' name existed in PBX.	N/A
Trunk's Emergency Outbound Caller ID	Optional	The maximum character length is 31. Numbers, letters, and characters `( ) . - + * #` are allowed.	N/A
Prepend for Emergency Number	Optional	The maximum character length is 10. Numbers, letters, and characters `( ) . - + * #` are allowed.	N/A

Related information

Export and Import Emergency Numbers
Import and Export -FAQ

Auto Provisioning Phone Information Parameters

Descriptions for parameters in exported and imported Auto Provisioning Phone Information CSV file.

Parameter	Description	Importance	Restriction	Default Value
Vendor	Phone vendor.	Required	Refer to the permitted value for the supported phone vendor.	N/A
Model	Phone model.	Required	Refer to the permitted value for the supported phone model of various phone vendors.	N/A
MAC Address	MAC address of the phone.	Required	Only numbers `0` - `9`, letters `A` - `Z` or `a` - `z`, hyphens `-` and colons `:` are allowed. The maximum character length is 17, where you can enter a combination of letters or numbers with a total of 12 characters. Examples are given below: • FA-B5-49-1A-2B-3C • FA:B5:49:1A:2B:3C • FAB5491A2B3C	N/A

Parameter	Description	Importance	Restriction	Default Value
Template	The configuration template used to provision the phone.	Required	The name of default or custom template. **Tip:** To obtain the value for this parameter, go to **Auto Provisioning > Resource Repository > Default Templates / Custom Templates**, then copy the name in the **Template Name** column.	N/A
Hot Desking Phone	Whether to set the phone as a hot desking phone.	Optional	Permitted value: • `0`: Disable. • `1`: Enable. **Note:** • **Hot Desking Phone** will be filled with default value `0` if you leave this field empty. • Do NOT set this parameter to `1` for DECT phones, or it will cause error when importing.	0
Authentication First-time	Whether to require users to enter authentication information on the phone before triggering the first-time provisioning.	Optional	**Note:** • Do NOT set this parameter to `1` for phones of **Grandstream**, **Alcatel-Lucent Enterprise**, and **Mitel**, or it will cause error when importing.	If **Hot Desking Phone** = `0`, the default value will be `1`. Otherwise, the default value will be `0`.

Parameter	Description	Importance	Restriction	Default Value
			• If **Hot Desking Phone** is set to `1`, this parameter MUST be `0`, or it will cause error when importing. Permitted value: • `0`: Disable. • `1`: Enable. **Note:** **Authentication First-time** will be filled with a default value if you leave this field empty.	
Assign Extension	The extension to be assigned to the phone.	Required if **Hot Desking Phone** = `0`.	**Note:** Do NOT set this parameter if **Hot Desking Phone** = `1`, or it will cause error when importing. Only numbers are allowed. For IP phones, the maximum character length is 7. For DECT phones, the maximum character length is 1024, and the format should be *`Handset Number`*:*`Extension Number`*. **Note:** Use `&` to separate multiple handsets. For example, `1:1002&2:1005&3:1006`.	N/A

Related information

[Export and Import Auto Provisioning Phone Information]
[Import and Export -FAQ]

Auto Provisioning Phone Information Parameters - Permitted Value

This topic provides the permitted values of the import and export Auto Provisioning phone information parameters, including Vendor and Model.

Vendor	Model
Yealink	• CP920, CP925 • SIP-CP935W, SIP-CP960, SIP-CP965 • SIP-T19P_E2, SIP-T21P_E2, SIP-T21_E2 • SIP-T23G, SIP-T23P, SIP-T27G, SIP-T29G • SIP-T30, SIP-T30P, SIP-T31, SIP-T31G, SIP-T31P, SIP-T31W, SIP-T33G, SIP-T33P, SIP-T34W • SIP-T40G, SIP-T40P, SIP-T41P, SIP-T41S, SIP-T41U, SIP-T42G, SIP-T42S, SIP-T42U, SIP-T43U, SIP-T44U, SIP-T44W, SIP-T46G, SIP-T46S, SIP-T46U, SIP-T48G, SIP-T48S, SIP-T48U • SIP-T52S, SIP-T53, SIP-T53W, SIP-T54S, SIP-T54W, SIP-T56A, SIP-T57W, SIP-T58, SIP-T58W • SIP-W60B, W70B, SIP-W75DM, SIP-W80DM, SIP-W90DM • VP59
Fanvil	• A10, A10W, A32, A32i, A320, A320i • FH-S01 • H1, H2U, H2U-V2, H3, H3W, H4, H4W, H5, H5W, H6, H6W • i10, i10D, i10S, i10SD, i10SV, i10V, i11S, i11SV, i12, i16S, i16SV, i16V, i18S • i20S, i23S • i30, i31S, i32V, i33V, i33VF • i504, i505, i506W, i507W, i51, i51W, i52, i52W, i53, i53W, i55A, i56A, i57A • i61, i62, i63, i64, i68 • PA2, PA2S, PA3 • V61G, V61W, V62, V62G, V62W, V63, V64, V65, V67 • W610W, W611W • X1S/X1SP, X1SG, X2, X2C, X210, X210-V2, X210i, X210i-V2 • X3SG, X3S, X3S(P) Lite, X3S(P) Pro, X3SW, X3SG Lite, X3SG Pro, X3U, X3U Pro, X301, X301G, X301W, X303, X303G, X303W, X305, X303-2 WIRE • X4, X4U, X4U-V2, X5U, X5U-V2, X5S • X6, X6U, X6U-V2 • X7, X7A, X7C, X7-V2, X7C-V2 • Y501, Y501W, Y501-Y, Y501W-Y
Grandstream	• GAC2500, GAC2570 • GRP2601, GRP2601P, GRP2602, GRP2602P, GRP2602G, GRP2602W, GRP2603, GRP2603P, GRP2604, GRP2604P

Vendor	Model
	• GRP2612, GRP2612P, GRP2612G, GRP2612W, GRP2613, GRP2614, GRP2615, GRP2616 • GRP2624, GRP2634, GRP2670 • GXP1610, GXP1620, GXP1625, GXP1628, GXP1630 • GXP2130, GXP2135, GXP2140, GXP2160, GXP2170
Htek	• UC902, UC902S, UC903 • UC912, UC912G, UC912E • UC921, UC921G, UC923, UC923U, UC924, UC924E, UC924U, UC924W, UC926, UC926E, UC926U • UCV10, UCV20, UCV50, UCV52, UCV53
Gigaset	• Gigaset N610 IP PRO, Gigaset N670 IP PRO • Gigaset N870 IP PRO, Gigaset N870 VI PRO • Maxwell Basic PRO • Maxwell 2 PRO, Maxwell 3 PRO, Maxwell 4 PRO
Snom	• HD100, HD101, HD350W, HD351W, HM201 • M100KLE, M500 • snomD120, snomD140, snomD150 • snomD315, snomD335, snomD385 • snomD713, snomD717, snomD735, snomD785 • snomD862, snomD865 • snomM300, snomM400, snomM900
Flyingvoice	• FIP10, FIP11C, FIP12WP, FIP13G, FIP14G, FIP15G, FIP15GPlus, FIP16, FIP16Plus • P10, P10P, P10G, P10W, P10LTE, P11, P11P, P11G, P11W, P11LTE • P20, P20P, P20W, P20G, P21, P21P, P21W, P22P, P22G, P23G, P23GW, P24G • flyphone • i86Box_Basic, i86Box_Indoor, i86Box_2Line, i86Box_PCBA, i86Box_NFC
Tiptel	• 3310, 3320, 3330, 3340
Alcatel-Lucent Enterprise	• H2, H2P, H3P, H3G, H6 • M3, M5, M7, M8
Mitel	• 6863i, 6865i, 6867i, 6869i, 6873i • 6920, 6930, 6940
Poly	• Edge_E100, Edge_E220,Edge_E300, Edge_E320, Edge_E350, Edge_E400, Edge_E450, Edge_E500, Edge_E550 • VVX_101, VVX_201, VVX_301, VVX_310, VVX_311, VVX_401, VVX_410, VVX_411, VVX_501, VVX_601, VVX_150, VVX_250, VVX_350, VVX_450
Wildix	• WP410R2, WP480R2, WP480R3, WP480R4, WP490R2, WP490R3

Trunk Parameters

Descriptions for parameters in exported and imported Trunk CSV file.

> **Note:**
> Only SIP Peer Trunk and Register Trunks can be exported and imported.

Parameter	Description	Importance	
Name	The trunk name.	Required	The maximum character length is Space and special characters are Trunk's name cannot be duplicate
Trunk Status	Whether to enable or disable the trunk.	Required	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Trunk Status** will be filled empty.
Trunk Type	Trunk type.	Required	Permitted value: • • *register* • *did_based* • *port_based* • *private_network* **Note:** • Importing Account • **Trunk Type**will be leave this field emp
Transport	The transport protocol that is provided by the ITSP.	Required	Permitted value: • If Trunk Type is *port_base* • If Trunk Type is *did_based* • If Trunk Type is `private_n` • If Trunk Type is `register`,

Parameter	Description	Importance	
			Note: **Transport** will be filled wi fields empty.
Hostname/IP	The IP address or the domain of the ITSP.	Required	The maximum character length is
Port	The trunk port.	Required	Only numbers between 0 and 655
Domain	The domain in SIP URI of a specific header like From, To header. **Note:** If the domain is not provided by ITSP, enter the same value as **Hostname/IP**.	Required	The maximum character length is
Username	The username to register to the ITSP.	Required if **Trunk Type** = *register*	The maximum character length is
Password	The password that is associated with the username.	Required if **Trunk Type** = *register*	The maximum character length is
Authentication Name	The authentication name to register to the ITSP.	Optional	The maximum character length is
Enable Outbound Proxy	Whether to enable or disable outbound proxy.	Required if **Trunk Type** = *register*	Permitted value: • *0*: Disable • *1*: Enable **Note:** **Enable Outbound Proxy** leave this field empty.
Outbound Proxy Server	The address of outbound proxy server.	Required if **Enable Outbound Proxy** = *1*	The maximum character length is
Port of Outbound Proxy Server	The port of outbound proxy server.	Required if **Enable Outbound Proxy** = *1*	Only numbers between 1 and 655
Codec Setting	The audio codec for trunk.	Required	Permitted value: *ulaw*, *alaw*, *g729*, *g726*, *speex*, *adpcm*, *vp8*, or *mpeg* For multiple Codec, please enter e.g. first_value1&second_value2.

Parameter	Description	Importance	
			Note: If the value you enter is no skipped.
DTMF Mode	The default mode for sending DTMF tones.	Required	Permitted value: `rfc4733`, `info`, `i` **Note:** **DTMF Mode** will be filled field empty.
DTMF FMTP	The value of the DTMF fmtp attribute when the DTMF mode is `rfc4733`.	Optional	Permitted value: `0-16`, `0-15`. **Note:** This field will be filled with empty.
Qualify	Whether to send SIP OPTION packet to check if the SIP device is up.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Qualify** will be filled with
Enable SRTP	Whether to enable or disable SRTP (encrypted RTP) for the trunk.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Enable SRTP** will be filled empty.
T.38 Support	Whether to enable or disable T.38 fax.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **T.38 Support** will be fille empty.

Parameter	Description	Importance	
Inband Progress	Whether to enable or disable inband progress.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Inband Progress** will be field empty.
Ignore 183 Message without SDP	Whether to ignore 183 messages without SDP.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Ignore 183 Message wit** if you leave this field empt
Dedicated Trunk	Whether it is a dedicated trunk that requires a dedicated internal IP address for connection.	Optional	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Dedicated Trunk** will be field empty.
Maximum Concurrent Calls	Specify the maximum number of concurrent calls allowed in the trunk.	Required	Only numbers are allowed. Specially, 0 means unlimited. The maximum character length is **Note:** **Maximum Concurrent C** leave this field empty.
Call Restriction Type	Specify based on which type of calls to define the maximum number of concurrent calls for this trunk.	Required	Permitted value: `outbound` or `all` **Note:** **Call Restriction Type** wi you leave this field empty.

Parameter	Description	Importance	
Default Outbound Caller ID	The caller ID that is displayed on the callee's device.	Optional	Numbers, letters, and characters The maximum character length is **Note:** The outbound caller ID sh
Default Outbound Caller ID Name	The caller ID name that is displayed on the callee's device.	Optional	The maximum character length is
Get Caller ID From	Decide from which header field will the trunk retrieve Caller ID.	Required	Permitted value: • `follow_system`: [Follow S • `from`: From • `contact`: Contact • `rpid`: Remote-Party-ID • `pai`: P-Asserted-Identity • `ppi`: P-Preferred-Identity **Note:** **Get Caller ID From** will b you leave this field empty.
Get DID From	Different devices or providers may contain DID numbers in different SIP headers. When an inbound call through a SIP trunk reaches the PBX, the PBX needs to retrieve a correct DID number, or the call will fail. Adjust the setting after analysis of the SIP packets sent from the trunk provider.	Required	Permitted value: • `follow_system`: [Follow S • `to`: To • `invite`: Invite • `diversion`: Diversion • `rpid`: Remote-Party-ID • `pai`: P-Asserted-Identity • `ppi`: P-Preferred-Identity • `pcpid`: P-Called-Party-ID **Note:** **Get DID From** will be fille leave this field empty.
From User Part	A From header contains caller ID and caller ID name. **From User Part** indicates caller ID.	Required	Permitted value: • `default`: [Default] • `ext_cid`: Extension Caller • `trunk_user`: Trunk Userna **Note:**

Parameter	Description	Importance	
			Only available whe • `trunk_def_outbcid`: Trun • `ext_outbcid`: Extension's • `outrounter_outbcid`: Ou • `originator_cid`: Originat • A customized value. **Note:** Fill in a desired val is 31. Only number are allowed. **Note:** **From User Part** will be fil this field empty.
From Display Name Part	A From header contains caller ID and caller ID name. **From Display Name Part** indicates caller ID name.	Required	Permitted value: • `default`: [Default] • `ext_cid_name`: Extension • `trunk_def_outbcid_name` • `ext_outbcid_name`: Exten • `originator_cid_name`: O • A customized value. **Note:** Fill in a desired val is 63. **Note:** **From Display Name Par** you leave this field empty.
Diversion	Define the parameters included in the Diversion SIP header.	Optional	Permitted value: • `default`: [Default] • `ext_cid`: Extension Caller • `trunk_user`: Trunk Userna **Note:** Only available whe

Parameter	Description	Importance	
			• *trunk_def_outbcid*: Trun • *ext_outbcid*: Extension's • *outrounter_outbcid*: Ou • *originator_cid*: Originat • A customized value. **Note:** Fill in a desired val is 31. Only number allowed. **Note:** Leave **Diversion** field em parameter with SIP INVIT
Remote-Party-ID	Define the parameters included in the Remote-Party-ID SIP header.	Optional	Permitted value: • *default*: [Default] • *ext_cid*: Extension Caller • *trunk_user*: Trunk Usern **Note:** Only available whe • *trunk_def_outbcid*: Trun • *ext_outbcid*: Extension's • *outrounter_outbcid*: Ou • *originator_cid*: Originat • A customized value. **Note:** Fill in a desired val is 31. Only number allowed. **Note:** Leave **Remote-Party-ID** this parameter with SIP IN
P-Asserted-Identity	Define the parameters included in the P-Asserted-Identity SIP header.	Optional	Permitted value: • *default*: [Default] • *ext_cid*: Extension Caller

Parameter	Description	Importance	
			• `trunk_user`: Trunk Userna **Note:** Only available whe • `trunk_def_outbcid`: Trun • `ext_outbcid`: Extension's • `outrounter_outbcid`: Ou • `originator_cid`: Originat • A customized value. **Note:** Fill in a desired valu is 31. Only number allowed. **Note:** Leave **P-Asserted-Identi** this parameter with SIP IN
P-Preferred-Identity	Define the parameters included in the P-Preferred-Identity SIP header.	Optional	Permitted value: • `default`: [Default] • `ext_cid`: Extension Caller • `trunk_user`: Trunk Userna **Note:** Only available whe • `trunk_def_outbcid`: Trun • `ext_outbcid`: Extension's • `outrounter_outbcid`: Ou • `originator_cid`: Originat • A customized value. **Note:** Fill in a desired valu is 31. Only number allowed. **Note:**

Parameter	Description	Importance	
			Leave **P-Preferred-Ident** this parameter with SIP IN
From Host Part	Define the domain or IP address to be used in the **From** field of a SIP INVITE header.	Required	**Note:** This parameter is only req Permitted value: • *default*: [Default] • A customized value. **Note:** Fill in a desired val is 255.
To Host Part	Define the domain or IP address to be used in the **To** field of a SIP INVITE header.	Required	**Note:** This parameter is only req Permitted value: • *default*: [Default] • A customized value. **Note:** Fill in a desired val is 255.
User Agent	If the ITSP requires User Agent for authentication, enter the User Agent information that is provided by the ITSP.	Optional	The maximum character length is
Realm	Realm is a string displayed to users so they know which username and password to use.	Optional	The maximum character length is
Send Privacy ID	Whether to send the Privacy ID in SIP header or not.	Required	Permitted value: • *0*: Disable • *1*: Enable
User Phone	Whether to add the parameter `user=phone` as a request line in	Required	Permitted value: • *0*: Disable • *1*: Enable

Parameter	Description	Importance	
	the header field of the SIP INVITE packet.		
100rel	Configure 100rel for this trunk.	Required	Permitted value: • *Required*: 100rel is require • *Supported*: 100rel is supp • *Disabled*: 100rel is disabl
Maxptime	Select the value of the Maxptime used when the PBX sends the INVITE packet.	Required	Permitted value: • *default*: PBX will send a the codec that is used for t • A customized value. **Note:** Fill in a desired val of 10 ranging from **Note:** **Maxptime** will be filled wi field empty.
Support P-Early-Media	Set whether the P-Early-Media field is included in the INVITE packet.	Required	Permitted value: • *0*: Disable • *1*: Enable
DID/DDI	A virtual number that is used to identify which path of the trunk is passing the call.	Required if **Trunk Type** = *register* or *did_based*	The maximum character length is **Note:** • If Trunk Type = reg characters ( ) . - • If Trunk Type = por letters, and charact
DID/DDI Name	The name of DID/DDI that is used to identify which path of the trunk is passing the call.	Optional	The maximum character length is

Related reference

Trunk DIDs/DDIs Parameters

Related information

Export and Import SIP Trunks
Import and Export -FAQ

Trunk DIDs/DDIs Parameters

Descriptions for parameters in exported and imported Trunk DIDs/DDIs CSV file.

Parameter	Description	Importance	Restriction	Default Value
DID/DDI	A virtual number that is used to identify which path of the trunk is passing the call.	Required if **Trunk Type** = *register* or *did_based*	The maximum character length is 31. **Note:** • If Trunk Type = *register* or *did_based*: Numbers, letters, and characters *( ) . - + *#* are allowed. • If Trunk Type = *port_based* or *private_network*: Numbers, letters, and characters *[ ] * # ( ) . - + !* are allowed.	N/A
DID/DDI Name	The name of DID/DDI that is used to identify which path of the trunk is passing the call.	Optional	The maximum character length is 127.	N/A

Related information

Export and Import Trunk DIDs/DDIs Numbers
Import and Export -FAQ

Trunk Outbound Caller ID Parameters

Descriptions for parameters in exported and imported Trunk Outbound Caller ID CSV file.

Parameter	Description	Importance	Restriction	Default Value
Create Method	The way to add outbound caller ID.	Required	Permitted value: • *single*: Shared Outbound Caller ID • *range*: Outbound Caller ID Range **Note:** **Create Method** will be filled with default value *single* if you leave this field empty.	single
Outbound Caller ID	The caller ID that is displayed on the callee's device for specific extensions.	Required	Numbers, letters, characters *[ ] ( ) . - + * #*, and placeholder *{{.Ext}}* are allowed. The maximum character length is 31(for each caller ID). For outbound caller id range: • Only numbers and character + (before numbers) are allowed. Fill in the start caller ID and the end caller ID with separator -, e.g. 5503301-5503310. • The start number and the end number must have the same amount of digits and both contain character + or neither. The range of start number and end number cannot exceed 500. • Then fill the extension range in Associated Extensions. The extension range and the outbound caller ID range must have the same amount of numbers.	N/A
Outbound Caller ID Name	The caller ID that is displayed on the callee's	Optional	The maximum character length is 127.	N/A

Parameter	Description	Importance	Restriction	Default Value
	device for specific extensions.			
Associated Extensions	The extensions that are associated with the Outbound Caller ID and Outbound Caller Name.	Required	Permitted value: one or more extension numbers and extension group names existed in PBX. • For multiple extensions or groups, please enter the numbers or names and use `&` as a separator, e.g. extension_number1&extension_number2&extension_group_name3. • If the extensions or groups you enter are not existing in PBX, it will be skipped. • For extension range, please fill in the start extension number and the end extension number with separator `-`, e.g. 1001-1010. The maximum number length is 7 (for each number).	N/A

Related information

Export and Import Trunk Outbound Caller IDs
Import and Export -FAQ

Inbound Caller ID Reformatting Rule Parameters

Descriptions for parameters in exported and imported 'Inbound Caller ID Reformatting Rule' CSV file.

Parameter	Description	Importance	Restriction	Default Value
Patterns	The inbound caller ID that matches this pattern will be reformatted.	Required	Numbers, letters, and characters `[ ] * # ( ) . - + !` are allowed. The maximum character length is 31.	N/A
Strip	Specify how many digits will be stripped from the beginning of the inbound caller ID.	Optional	Only numbers are allowed. The maximum character length is 2.	N/A

Parameter	Description	Importance	Restriction	Default Value
Prepend	Specify the digits that will be prepended to the inbound caller ID.	Optional	Numbers, letters, and characters ( ) . - + * # are allowed. The maximum character length is 31.	N/A

Related information

Export and Import Inbound Caller ID Reformatting Rules
Import and Export -FAQ

DID Number to Specific Extension Parameters

Descriptions for parameters in exported and imported 'DID Number to Specific Extension' CSV file.

Parameter	Description	Importance	Restriction	Default Value
DID Number	The specific DID number that is used to match the callee number. Only when the callee number is equal to the DID number will the inbound call go through the inbound route.	Required if **DID Matching Mode** = `pattern_to_ext_list`	The maximum character length is 31. Only numbers, letters, and characters `( ) . - + * #` are allowed.	N/A
Extension Matched	The destination of the specific DID number. The DID number and the extension is in one-to-one correspondence.	Required if **DID Matching Mode** = `pattern_to_ext_list`	Permitted value: Extension numbers	N/A

Related information

Import and Export -FAQ

Inbound Route Parameters

Descriptions for parameters in imported and exported Inbound Route CSV file.

Parameter	Description	Importance	Restriction	Default Value
Name	The name of inbound route.	Required	Space and special characters are not allowed. Inbound route's name cannot be duplicated. The maximum character length is 63.	N/A
Inbound Alert Info	The Alert Info field is used to configure distinctive ring tones for incoming calls.	Optional	Only numbers and letters are allowed. The maximum character length is 31.	N/A
DID Matching Mode	The DID matching mode.	Optional	Permitted value: • *patterns*: DID Patterns • *pattern_to_ext*: DID Pattern to Extensions • *range_to_ext*: DID Range to Extension Range • pattern_to_ext_list: DID Number to Specific Extension **Note:** **DID Matching Mode** will be filled with default value *patterns* if you leave these fields empty.	patterns
DID Pattern	The DID pattern that is used to match callee number. Only when the callee number is matched will the inbound call go through this route.	Required if **DID Matching Mode** ≠ *patterns*	• If **DID Matching Mode** = *patterns*, you can enter one or more patterns. Numbers, letters, and characters *[ ] * # ( ) . - + !* are allowed. The maximum character length is 31 (for each DID). Please use *&* as a separator for multiple patterns, e.g. pattern1&pattern2.	N/A

Parameter	Description	Importance	Restriction	Default Value
			• If **DID Matching Mode** = *pattern_to_ext*, only numbers, letters *X Z N*, characters *[ ] * # - +*, and placeholder *{{.Ext}}* are allowed. The maximum character length is 31. The **Default Destination** must be *pattern_to_ext*, then fill multiple extension numbers with separator *&* in Number. • If **DID Matching Mode** = *range_to_ext*, only numbers and character + (before numbers) are allowed. The maximum character length is 16 (for each DID). Please enter the start DID and the end DID with separator -, e.g. 5503301-5503305. The **Default Destination** must be *range_to_ext*, then fill the start number and the end number with separator - in Number, e.g. 1001-1005.	
Caller ID Matching Mode	The Caller ID matching mode.	Required	Permitted value: • *patterns*: Caller ID Matching Settings • *phonebook*: Match Contacts' Caller ID in Specific Phonebooks **Note:** **Caller ID Matching Mode** will be filled with default value *patterns* if you leave this fields empty.	patterns

Parameter	Description	Importance	Restriction	Default Value
Caller ID Pattern	The pattern used to match caller ID. Only when the caller ID matches the pattern can user dials in through this route.	Optional	• If **Caller ID Matching Mode** = *patterns*, the maximum character length is 31 (for each pattern). Numbers, letters, characters *[ ] * # ( ) . - + !* are allowed. For multiple patterns, enter patterns and use *&* as a separator, e.g. pattern1&pattern2. • If **Caller ID Matching Mode** = *phonebook*, the permitted value is one or more phonebook names existed in PBX. For multiple phonebook names, enter names and use *&* as a separator, e.g. name1&name2. If the phonebook you enter does not exist in PBX, it will be skipped.	N/A
Trunks	The trunks that incoming calls will be routed by this inbound route. The PBX will route inbound calls through this route when external users call the selected trunk number.	Required	Permitted value: one or more trunk names existed in PBX. For multiple trunks, please enter trunk names and use *&* as a separator, e.g. name1&name2. **Note:** If the trunks you enter are not existing in PBX, Trunks will be skipped.	N/A
Default Destination	The default destination to receive inbound calls.	Required	Permitted value: • *end_call*: Hang up • *extension*: Extension • *range_to_ext*: Match extension Range (DID Matching Mode = range_to_ext)	end_call

Parameter	Description	Importance	Restriction	Default Value
			• *pattern_to_ext*: Match selected Extension (DID Matching Mode = pattern_to_ext) • pattern_to_ext_list: DID number match specific extension (DID Matching Mode = pattern_to_ext_list) • *ext_vm*: Extension Voicemail • *group_vm*: Group Extension • *ivr*: IVR • *ring_group*: Ring Group • *queue*: Queue • *conference*: Conference • *fax_to_email*: Fax to Email **Note:** **Default Destination** will be filled with default value *end_call* if you leave these fields empty.	
Number of Default Destination	The destination number to receive inbound calls.	Required if **Default Destination** ≠ *end_call* or *pattern_to_ext_list*	Permitted value: • If **Default Destination** = *Extension*, *Extension Email*, *Group Voicemail*, *IVR*, *Ring Group*, *Queue*, *Conference*, or *Fax to Email*, please fill in a number. • If **Default Destination** = *Match extension Range*, please fill in a range of extension, e.g. 1000-1010. The maximum number length is 7 (for each number). • If **Default Destination** = *Match selected Extension*, please fill in numbers or names and use *&* as a separator, e.g. extension_number1&extensi	N/A

Parameter	Description	Importance	Restriction	Default Value
			on_number2&extension_group_name3. **Note:** If the numbers or names you enter are not existing in PBX, Number of Default Destination will be skipped.	
Enable Fax Detection	Whether to enable or disable FAX detection.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Enable Fax Detection** will be filled with default value `0` if you leave this field empty.	0
Fax Destination	The destination to receive fax.	Required if **Enable Fax Detection** = `1`	Permitted value: • `end_call`: Hang Up • `extension`: Extension • `fax_to_email`: Fax to Email **Note:** **Fax Destination** will be filled with default value `extension` if you leave this field empty.	extension
Number of Fax Destination	The destination number to receive fax.	Required if **Fax Destination** ≠ `end_call`	Permitted value: extension numbers existed in PBX. • If **Fax Destination** = `Extension`, fax will be sent to extension number. • If **Fax Destination** = `Fax to Email`, fax will be sent to extension's email address.	N/A

Related reference

DID Number to Specific Extension Parameters

Related information

Export and Import Inbound Routes

Import and Export -FAQ

Outbound Route Parameters

Descriptions for parameters in exported and imported Outbound Route CSV file.

Parameter	Description	Importance	Restriction	Default Value
Name	The name of outbound route.	Required	Space and special characters are not allowed. Outbound route's name cannot be duplicated. The maximum character length is 63.	N/A
Outbound Caller ID	The caller ID that is displayed on the callee's device.	Optional	Numbers, letters, and characters `[ ] ().- + * #` and placeholder `{{.Ext}}` are allowed. The maximum character length is 31.	patterns
Pattern	The pattern used to match a callee number. Only when the callee number is matched will the outbound call go through this route.	Required	Numbers, letters, and characters `[ ] * # ( ) . - + !` are allowed. The maximum character length is 31. **Note:** **Pattern** will be filled with default value `X.` if you leave these fields empty.	X.
Strip	The number of digits that will be stripped from the front of callee number before the call is placed.	Optional	Only numbers 1 - 16 are allowed.	N/A

Parameter	Description	Importance	Restriction	Default Value
Prepend	The digits that will be prepended to the callee number before the call is placed.	Optional	Numbers, letters, and characters `( ) . - + * #` are allowed. The maximum character length is 31.	N/A
Trunks	The trunks that can be used to dial out. The PBX will route outbound calls through this trunk when the dialed number matches the outbound route.	Required	Permitted value: one or more trunk names existed in PBX. For multiple trunks, please enter trunk names and use `&` as a separator, e.g. name1&name2. **Note:** If the trunk you enter does not exist in PBX, it will be skipped.	N/A
Rrmemory Hunt	Whether to remember which trunk was used last time, and then use the next available trunk to call out.	Required	Permitted value: • `0`: Disable • `1`: Enable **Note:** **Rrmemory Hunt** will be filled with default value `0` if you leave this field empty.	0
Extensions	The extensions that are allowed to make outbound calls through this route.	Optional	Permitted value: one or more extension numbers, extension group names, or organization names existed in PBX. **Format**: • When entering the organization name, you must enter the complete path with parent organization(s) connected by character `/`, and with the prefix `Organization_`, i.e. `Organization_{Parent Organization}/{Organization Name}`. For example,	extension

Parameter	Description	Importance	Restriction	Default Value
			`Organization_Yeastar/Marketing Center/Training Team`. • For multiple extensions, extension groups, or organizations, please enter the numbers or names and use *&* as a separator, e.g. extension_number1&extension_number2&extension_group_name3&Organization_Parent Organization/Organization Name4. **Note:** If the extensions, groups, or organizations you entered are not existing in PBX, it will be skipped.	
Outbound Route Password	Whether to require users to enter the same PIN to make outbound calls through this route.	Required	Permitted value: *disable*, *single_pin*, or *pin_list*. **Note:** **Outbound Route Password** will be filled with default value *disable* if you leave these fields empty.	disable
PIN	The PIN is required to make outbound calls through this route.	Required if **Outbound Route Password** is *single_pin*	Only numbers are allowed. The minimum character length is 3 and the maximum is 15.	N/A
PIN List	The PIN codes in the selected PIN list are required to make	Required if **Outbound Route**	Permitted value: The name of a PIN list existed in PBX.	N/A

Parameter	Description	Importance	Restriction	Default Value
	outbound calls through this route.	**Password** is *pin_list*	**Note:** If the PIN list name you entered is not existing in PBX, it will be skipped.	

Related information

Export and Import Outbound Routes
Import and Export -FAQ

Outbound Call Frequency Restriction Rule Parameters

Descriptions for parameters in exported and imported 'Outbound Call Frequency Restriction' CSV file.

Table 49.

Parameter	Description	Importance	Restriction	Default Value
Name	The name of Outbound Call Frequency Restriction rule.	Required	The maximum character length is 127. **Note:** The name of an Outbound Call Frequency Restriction cannot be duplicated.	N/A
Restrictions	How many outbound calls users can make within a specific time period.	Required	**Format**: *{number_of_calls_limit}*/*{time_limit}*/*{time_unit}* **Example**: `200/10/second` **Note:** Use *&* to separate multiple restrictions, e.g. `200/10/second&3000/1/minute`. **Variables**: *{number_of_calls_limit}*:	N/A

Table 49. (continued)

Parameter	Description	Importance	Restriction	Default Value
			• Only numbers are allowed. • The maximum character length is 5. *{time_limit}*: • Only numbers are allowed. • The maximum character length is 5. *{time_unit}*: Permitted value: *second* or *minute*.	

Related information

Export and Import 'Outbound Call Frequency Restriction' Rules

Import and Export -FAQ

Rate Parameters

Descriptions for parameters in exported and imported Rate CSV file.

Parameter	Importance	Restriction	Default Value
Name	Required	The maximum character length is 127.	N/A
Match Prefix	Optional	The maximum character length is 31. Numbers, letters, and characters *[]*#().-+!* are allowed.	N/A
Number Length	Optional	The maximum character length is 2. Only numbers are allowed.	N/A
Rate	Required	The maximum character length is 7. Only numbers and characters . are allowed.	0
Billable Unit(s)	Required	The maximum character length is 3. Only numbers are allowed.	60
Initial Time(s)	Required	The maximum character length is 3. Only numbers are allowed.	0
Initial Cost	Required	The maximum character length is 7.	0

Parameter	Importance	Restriction	Default Value
		Only numbers and characters `.` are allowed.	

Related information

Export and Import Call Rate Rules
Import and Export -FAQ

Allowed Numbers Parameters

Descriptions for parameters in exported and imported Allowed Numbers CSV files.

Table 50.

Parameter	Importance	Restriction	Default Value
Name	Required	The maximum character length is 127.	N/A
Type	Required	Permitted value: • *inbound*: Allow the number(s) to call into the PBX. • *outbound*: Allow PBX extensions to call the number(s). • *both*: Allow the number(s) to call into the PBX and allow PBX extensions to call the number(s). **Note:** **Type** will be filled with default value *inbound* if you leave this field empty.	inbound
Number	Required	Numbers, letters, and characters `[ ] * # ( ) . - + !` are allowed. For multiple numbers or number patterns, use *&* as a separator, eg. number1&number2.	N/A

Related information

Import and Export -FAQ

Blocked Numbers Parameters

Descriptions for parameters in exported and imported Blocked Numbers CSV files.

Table 51.

Parameter	Importance	Restriction	Default Value
Name	Required	The maximum character length is 127.	N/A
Type	Required	Permitted value: • *inbound*: Block the number(s) from calling into the PBX. • *outbound*: Block the PBX extensions from calling the number(s). • *both*: Block the number(s) from calling into the PBX and block the PBX extensions from calling the number(s). **Note:** **Type** will be filled with default value *inbound* if you leave this field empty.	inbound
Number	Required	Numbers, letters, and characters `[ ] * # ( ) . - + !` are allowed. For multiple numbers or number patterns, use *&* as a separator, eg. number1&number2.	N/A

Related information

Import and Export -FAQ