Admin Guide

Yeastar Cloud PBX

Version: 81.13.0.15
Date: 2020-03-10

Support: +86-592-5503301
Support: support@yeastar.com
https://www.yeastar.com

Yeastar Information Technology Co.Ltd.

Contents

Admin Guide

Admin Guide for Yeastar Cloud PBX.

About this guide

In this guide, we describe every detail on the functionality and configuration of the Yeastar Cloud PBX. We begin by assuming that you are familiar with networking and other IT disciplines.

Audience

This guide is for administrators who need to prepare for, configure and operate Yeastar Cloud PBX.

Extensions

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

SIP Extension

A SIP extension is based on SIP protocol.

To use a SIP extension, you need to enter the extension credentials on an IP phone or a softphone. After the extension is registered on a phone, you can make and receive calls.

Extension format

Yeastar Cloud PBX supports 1-digit to 7-digit extension format. The default extension format is 4-digit number.

Before you create extensions, you can go to **Settings > PBX > General > Preferences > Extension Preferences > User Extension** to change the extension format and range.

Extension Basic Setup

Create Extensions

Extension Creation Overview

Yeastar Cloud PBX supports SIP forking, which enables an extension number to register on multiple SIP phones simultaneously.

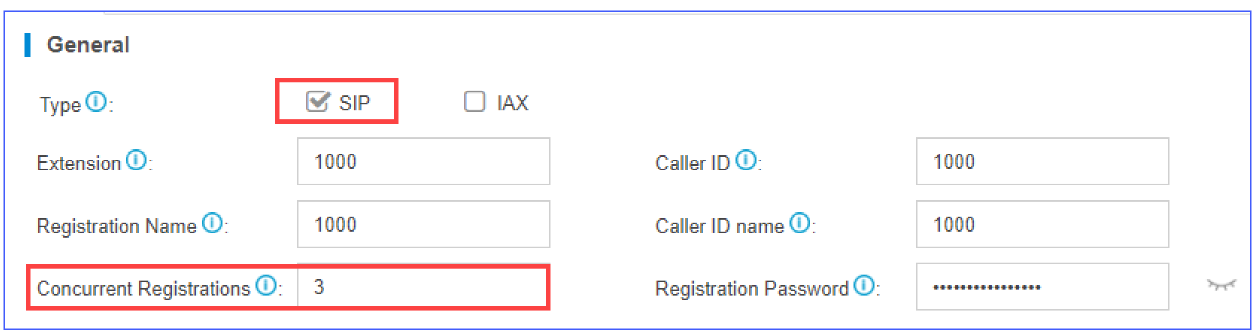
SIP Forking

Yeastar Cloud PBX supports SIP forking, which enables an extension number to be registered by multiple SIP phones. When a call reaches the extension, all registered phones will ring simultaneously, and you can take the call from any device easily.

You can configure SIP Forking on the extension configuration page. The value of **Concurrent Registrations** limits how many SIP phones the extension can be registered on.

Note:

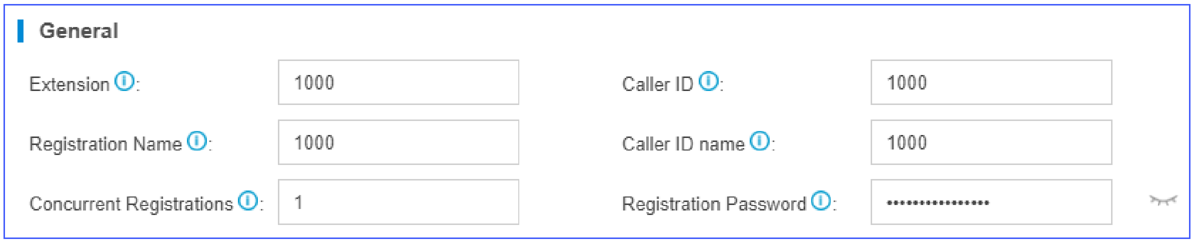
- The limit of concurrent registrations is **4**.
- By default, if one SIP phone is busy, other SIP phones still can receive calls when calls reach the extension. To restrict other phones from receiving calls when the extension is busy, you can enable **All Busy Mode for SIP Forking** (**Settings > PBX > General > SIP > Advanced**).



Create an Extension

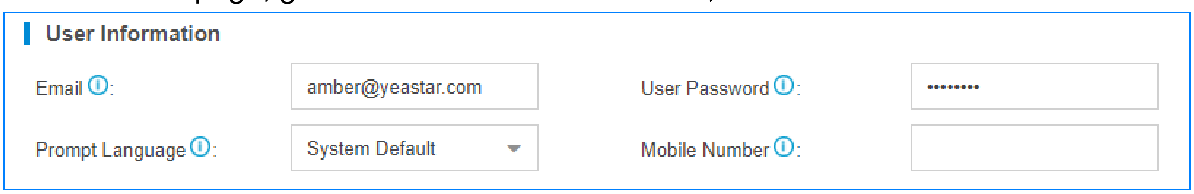
Before registering a SIP account on phones, you need to create a SIP account.

1. Go to **Settings > PBX > Extensions**, click **Add**.
2. On the **Basic** page, go to **General** section, and set the general settings of the extension.



- **Extension**: Enter the extension number.
- **Caller ID**: Enter the caller ID number. The called party will see this caller ID number when the extension user makes an outgoing call.
- **Registration Name**: The name used to register a SIP extension.
- **Caller ID name**: Enter the caller ID name. The called party will see this caller ID name when the extension user makes an outgoing call.
- **Concurrent Registrations**: Yeastar Cloud PBX supports to register one extension number on multiple phones. When a call reaches the extension number, all phones will ring. The maximum number of concurrent registrations is 4.
- **Registration Password**: The password is used to register the extension.

3. On the **Basic** page, go to **User Information** section, and set the user information.



- **Email**: Extension user can reset his/her login password, receive voice mails, faxes, or PBX notifications via this email address.
- **User Password**: The password is used to log in the PBX or log in Linkus mobile client. The password is generated randomly by default.
- **Prompt Language**: The default prompt language is the same as the system language. If the extension user speaks foreign language, you can set a specific system prompt.

  Note: Before selecting other system prompts, go to **Settings > PBX > Voice Prompts > System Prompt** to download online prompts.

- **Mobile Number**: Extension user can receive the PBX notifications or forwarded calls on this mobile number.

4. **Optional:** Click **Presence**, **Features**, **Advanced**, or **Call Permission** tab to configure other settings.

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

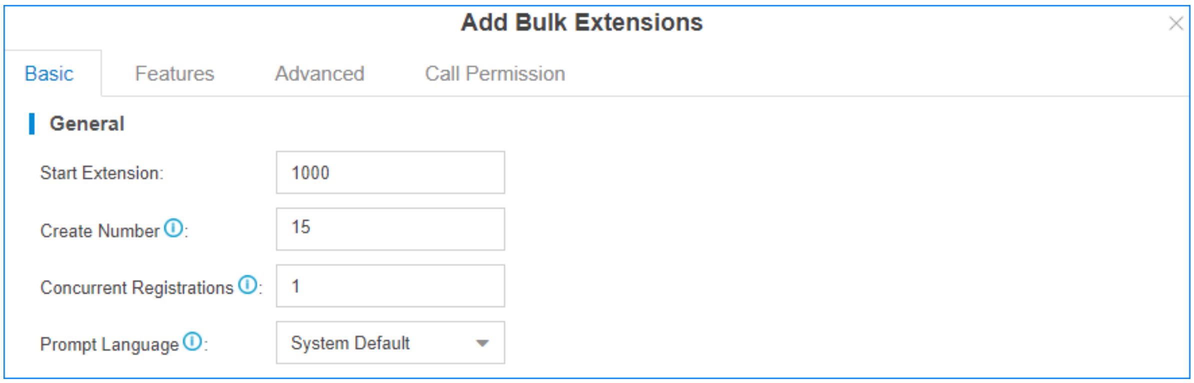
Bulk Create Extensions

Yeastar Cloud PBX supports to add SIP extensions in bulk.

1. Go to **Settings > PBX > Extensions**, click **Bulk Add**.
2. On the **Basic** page, go to **General** section, and configure the following settings:

   Note:

   - A random **Registration Password** and a random **User Password** will be assigned for each extension.

   - **Start Extension**: Enter the first extension number. The system will create extensions in bulk starting with the extension number.
   - **Create Number**: Enter the number of extensions that will be created.
   - **Concurrent Registrations**: Yeastar Cloud PBX supports to register one extension number on multiple phones. When a call reaches the extension number, all phones will ring.
   - **Prompt Language**: The language of voice prompts. The default prompt language is the same as the system language. If the extension user speaks foreign language, you can set a specific system prompt.

     Note: Before selecting other system prompts, go to **Settings > PBX > Voice Prompts > System Prompt** to download online prompts.

3. **Optional:** Click **Features**, **Advanced**, or **Call Permission** tab to configure other settings.
4. Click **Save** and **Apply**.

Related tasks

[Bulk Edit Extension Names and Emails](#)

Related information

[Register a SIP Extension](#)

Register Extensions

Register a SIP Extension

To make calls and receive calls from a SIP extension, you need to register the SIP extension on an IP phone or soft phone.

1.Gather information of extension registration

For most SIP phones, the following items are needed for the SIP phone to register with Yeastar Cloud PBX.

- Domain name of PBX
- SIP registration port: The default port is 5060 on Yeastar Cloud PBX.
- Extension information
  - Extension Number
  - Registration Name
  - Registration Password
  - Caller ID Name
  - Transport

2.Register the extension on a phone

Log in the phone web interface, fill in and save the required items to register the SIP extension.

3.Confirm registration status

You can do one of the followings to check if the extension is registered.

- On the phone web interface, check if the status indicates that the extension is registered.
- Log in PBX web interface, go to **PBX Monitor > Extensions** to check if the status shows .

Related tasks

Register Yealink Phone with Yeastar Cloud PBX
Register Htek Phone with Yeastar Cloud PBX
Register Cisco Phone with Yeastar Cloud PBX
Register Fanvil Phone with Yeastar Cloud PBX
Register Snom Phone with Yeastar Cloud PBX

Manage Extensions

Change Extension Range

The default extension range is from 1000 to 5999. Before you create extensions, you can change the extension range according to your needs.

1. Log in PBX web interface, go to **Settings > PBX > General > Preferences > Extension Preferences**.
2. Change the range of **User Extensions**.
3. Click **Save** and **Apply**.

Edit Extensions

After creating extensions, you may need to change extension settings. You can edit an extension, or edit extensions in bulk.

Edit an Extension

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. On **Extensions** page, click beside the extension that you want to edit.
3. Change extension settings according to your needs.
4. Click **Save** and **Apply**.

Bulk Edit Extensions

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. On **Extensions** tab, select the checkbox of desired extensions, and click **Edit**.
3. Change extension settings according to your needs.
4. Click **Save** and **Apply**.

Bulk Edit Extension Names and Emails

To bulk edit the extension names and emails, you need to export the extensions from Yeastar Cloud PBX first, edit the extension names and email addresses in the CSV file, then import the file to the PBX.

1. Log in PBX web interface, go to **Settings > PBX > Extensions**, click **Export** to export all the extensions.
2. Edit the CSV file, enter the users' names and email addresses, then save the file.

	A	B	C	D	E	F	G	H	I	J	K	L
1	type	username	fullname	callerid	registerr	registerp	loginpass	vmsecret	hasvoicem	enablevm	email	ringtimec
2	SIP	1000	carol	1000	1000	XbY-?01S_@NWOYPP		1000	yes	no	carol@yeastar.com	30
3	SIP	1001	eve	1001	1001	tIf?1@YjretXYPVY		1001	yes	no	eve@yeastar.com	30
4	SIP	1002	ina	1002	1002	??F-52ivj745omnr		1002	yes	no	ina@yeastar.com	30
5	SIP	1003	apple	1003	1003	k1QCFN-˜GOUWTARO		1003	yes	no	apple@yeastar.com	30
6	SIP	1004	david	1004	1004	3kGSY@@˜?onxJM7O		1004	yes	no	david@yeastar.com	30
7	SIP	1005	amber	1005	1005	_4Q3-a˜C40INC_0P		1005	yes	no	amber@yeastar.com	30
8	SIP	1006	alan	1006	1006	i_TU_G2J˜_˜@ˆYFP		1006	yes	no	alan@yeastar.com	30
9	SIP	1007	jason	1007	1007	@*?4rF*-S1*M_HKG		1007	yes	no	jason@yeastar.com	30
10	SIP	1008	ramon	1008	1008	@-N81AlTKIGIXJTE		1008	yes	no	ramon@yeastar.com	30
11	SIP	1009	harry	1009	1009	?*0es*tuGIN@-hsg		1009	yes	no	harry@yeastar.com	30
12	SIP	1010	pixy	1010	1010	D*2-*_to16408512		1010	yes	no	pixy@yeastar.com	30
13	SIP	1011	rose	1011	1011	ˆF2?65otv2plerrj		1011	yes	no	rose@yeastar.com	30
14	SIP	1012	hermy	1012	1012	@T1u*?1UG_˜KsrVR		1012	yes	no	hermy@yeastar.com	30
15	SIP	1013	gary	1013	1013	W˜h-˜6x??-ˆ?_?ˆ_		1013	yes	no	gary@yeastar.com	30
16	SIP	1014	jerry	1014	1014	712rx_?BUAmobLLG		1014	yes	no	jerry@yeastar.com	30
17												

- **fullname**: Enter the user's name. The **fullname** stands for the **Caller ID Name**.
- **email**: Enter the user's email address.

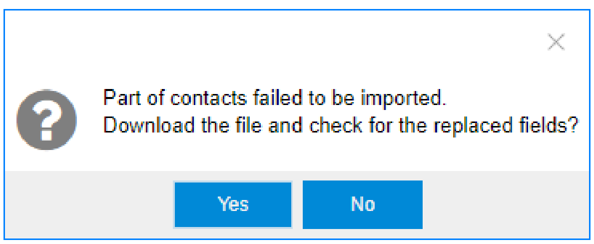
3. Import the CSV file to the PBX.

    a. Go to **Settings > PBX > Extensions**, click **Import**.

    b. On the pop-up dialog, click **Browse**, select your CSV file.

    c. Click **Import**.

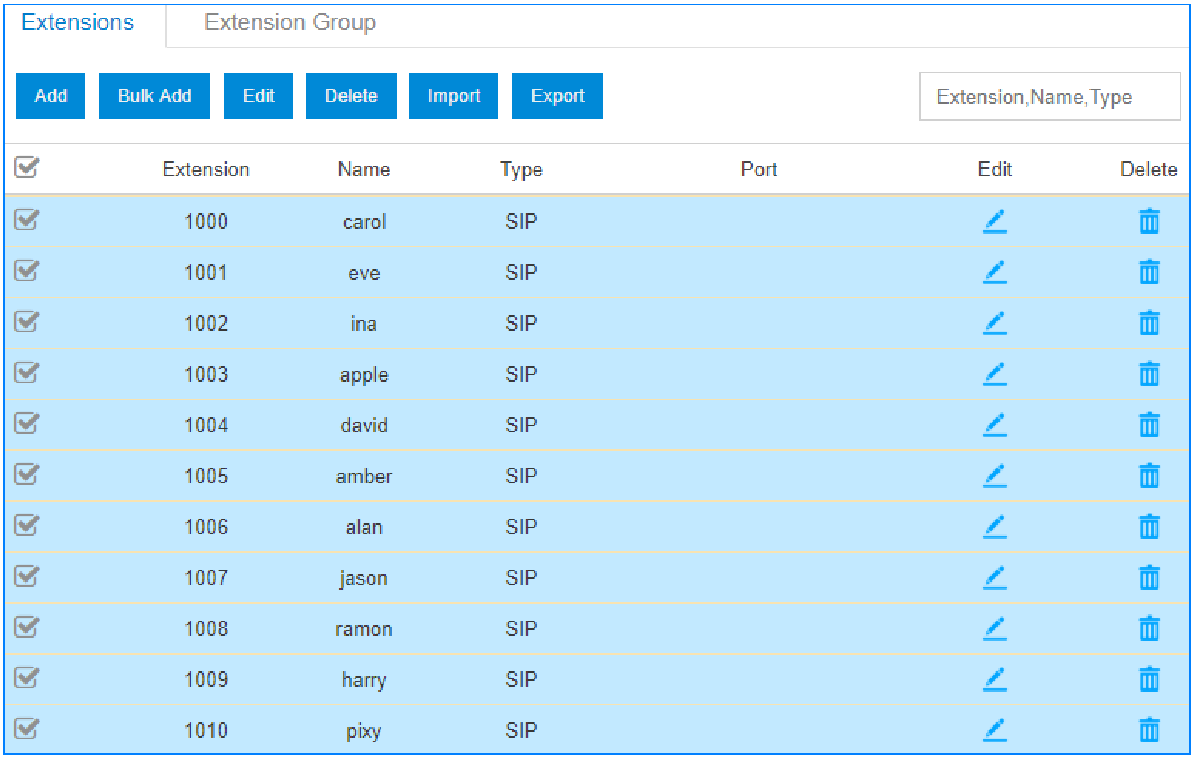
    Note: You may get an error prompt like the following figure.



    d. Click **Yes** to check the log.

    Note: Ignore the error if the Error Cause displays "username[*1000*]: The imported record is existing, the record has been overwritten".

4. Check the imported extensions on your PBX.



	Extension	Name	Type	Port	Edit	Delete
	1000	carol	SIP			
	1001	eve	SIP			
	1002	ina	SIP			
	1003	apple	SIP			
	1004	david	SIP			
	1005	amber	SIP			
	1006	alan	SIP			
	1007	jason	SIP			
	1008	ramon	SIP			
	1009	harry	SIP			
	1010	pixy	SIP			

Delete Extensions

When an employee leaves or an extension is no longer needed, you can delete the extension from the Yeastar Cloud PBX.

Delete an Extension

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. On **Extensions** page, click beside the extension that you want to delete.
3. Click **Save** and **Apply**.

Bulk Delete Extensions

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. On **Extensions** page, select the checkbox of desired extensions, and click **Delete.**
3. Click **Save** and **Apply**.

Import/Export Extensions

The extensions configured on Yeastar Cloud PBX can be exported and saved as a template. You can fill in desired extension information and import the CSV file to PBX again.

Export Extensions

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. Click **Export** to export the extensions to a CSV file.

Import Extensions

Tip: You can export extensions first, and use the CSV file as a template.

1. Log in PBX web interface, go to **Settings > PBX > Extensions**.
2. Click **Import**.
3. On the **Import Extension** page, click **Browse** to select your CSV file.
4. Click **Import**.

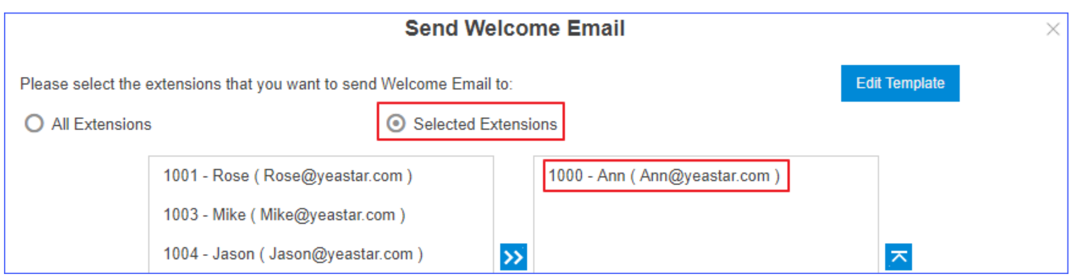
Send Extension Information

After finishing the extension configurations, you can send the extension information to the extension users' emails. The extension users can get their extension registration information, Linkus login information, PBX login information in the email.

1. Log in PBX web interface, go to **Settings > PBX > Extensions**, click **Welcome Email**.

2. Select the extensions that you want to send Welcome Email to.
    - To send emails to all extensions, select **All Extensions**.
    - To send emails to specific extensions, follow the instruction.

        a. Select **Selected Extensions**.
        b. Select the desired extensions from **Available** box to **Selected** box.



3. Click **Send**.

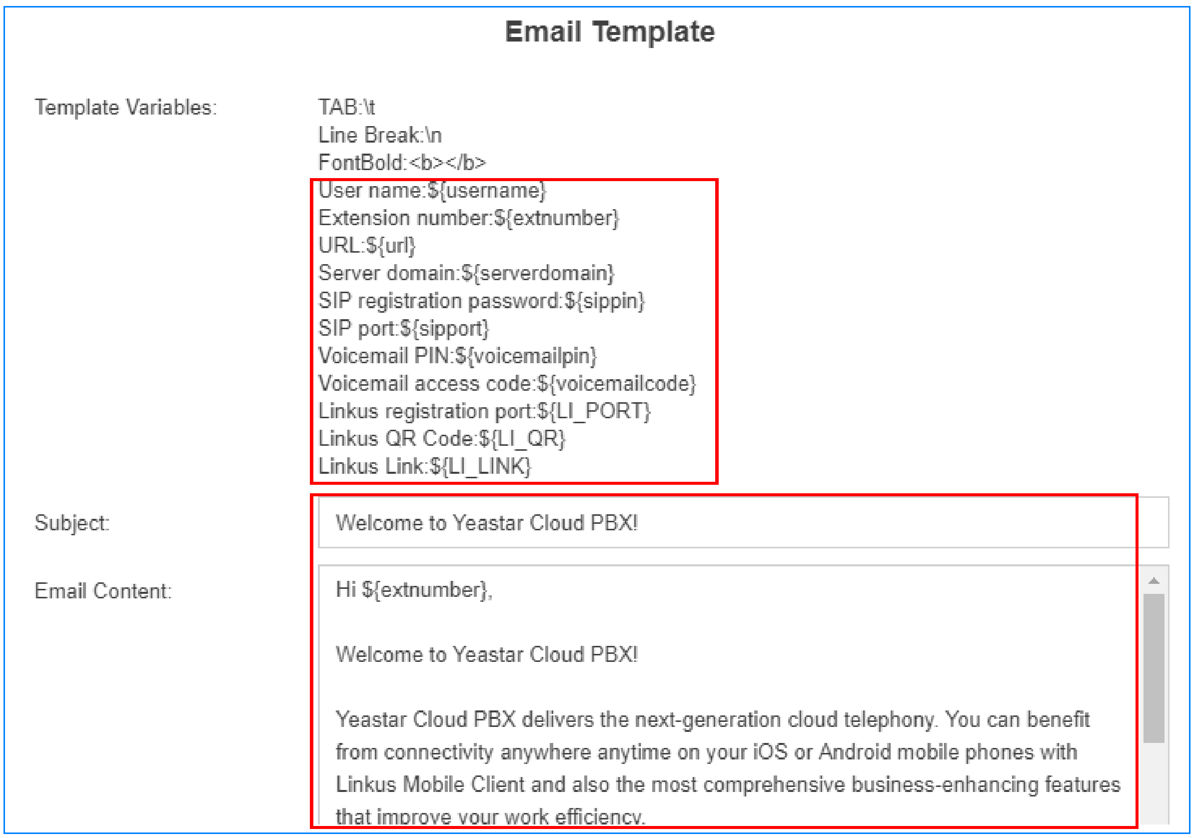
Related tasks
[Edit 'Welcome Email' Template]

Edit 'Welcome Email' Template

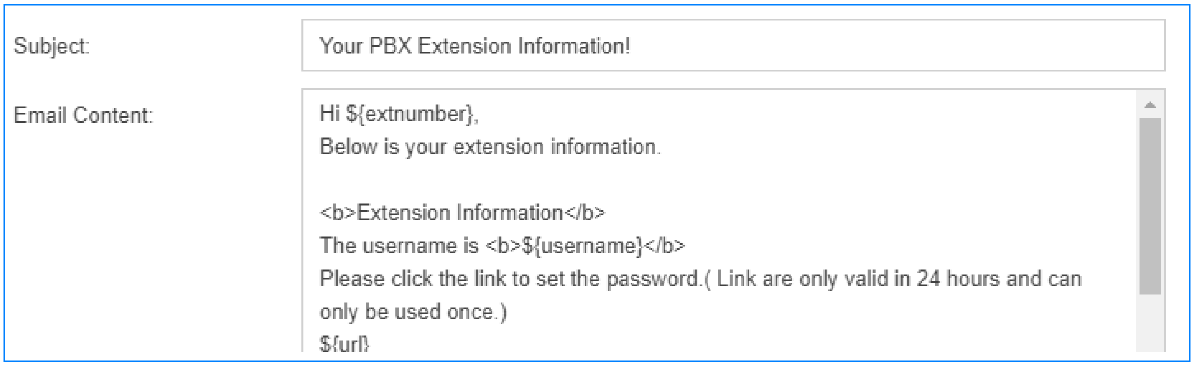
Yeastar Cloud PBX has a default template of the welcome email. You can change the email subject and email contents according to your needs.

1. Log in PBX web interface, go to **Settings > PBX > Extensions**, click **Welcome Email**.
2. On the **Send Welcome Email** page, click **Edit Template**.
   You will see the description of variables and the default email contents.

   Note: The variables in the email contents are unchangeable.



3. Edit the email subject and email contents.



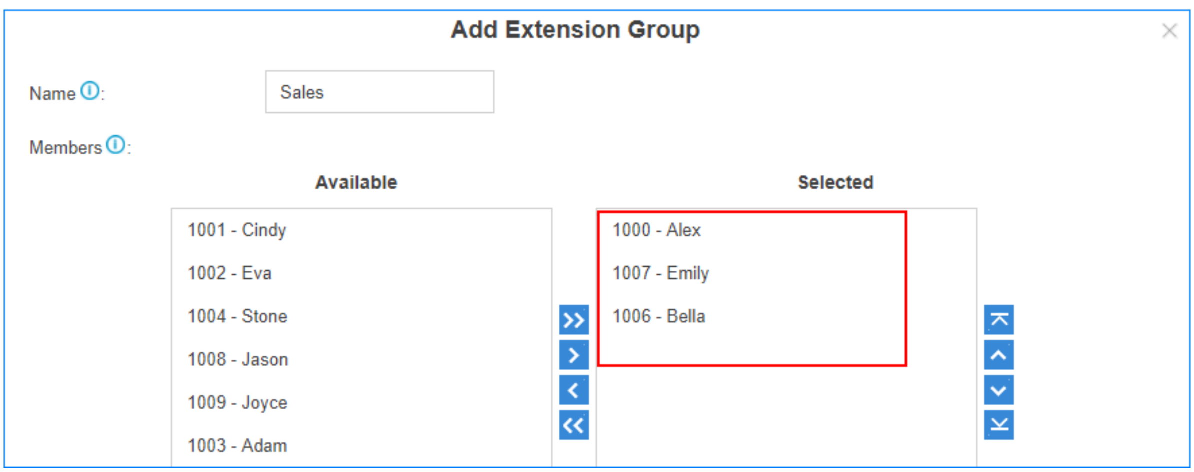
4. Click **Save** and **Apply**.

Extension Groups

Create an Extension Group

You can assign and categorize extensions in different groups. Extension groups simplify the configuration process.

1. Go to **Settings > PBX > Extensions > Extension Group**, click **Add**.
2. Set the **Name** to help you identify the group.
3. In the **Available** box, select the extensions to the **Selected** box.



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

Manage Extension Groups

Edit extension groups

You can edit the group name, add more extensions to the group or remove extensions from the group.

1. Go to **Settings > PBX > Extensions**, search and find the desired extension group, click .
2. Edit the group as you need.
3. Click **Save** and **Apply**.

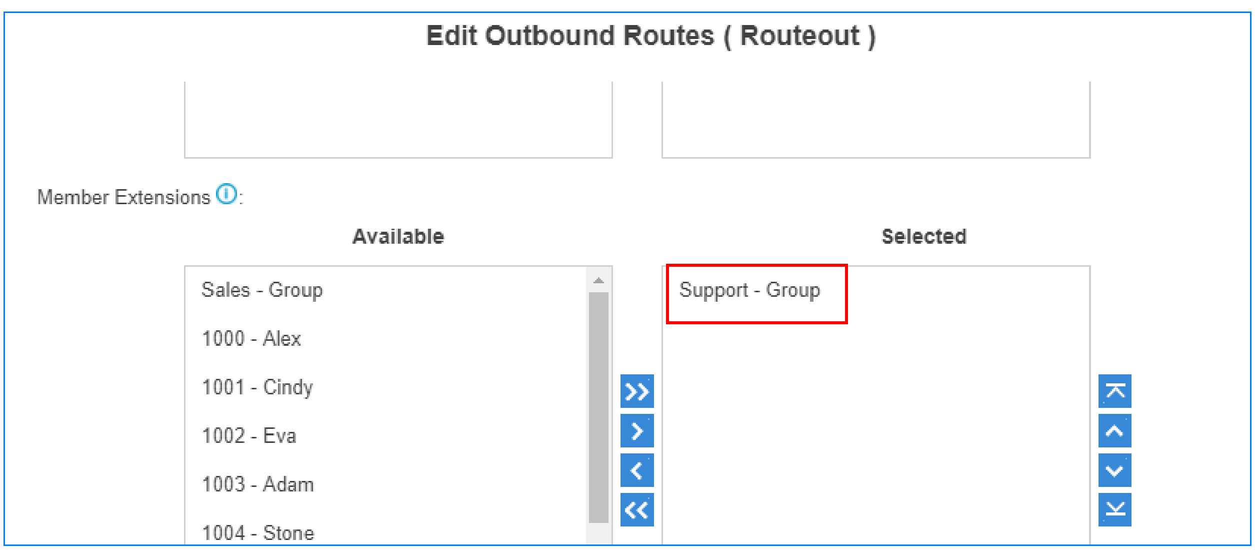
Delete extension group

1. Go to **Settings > PBX > Extensions > Extension Group**, search and find the desired extension group, click .
2. Click **Yes** to confirm the deletion.

Extension Groups Application

You can use the extension groups when you need to assign extensions for outbound routes, ring groups, queues, etc.

For example, you need to set an outbound route and only allow the Support group members to make outbound calls through this route. You can simply assign the Support extension group instead of assigning an extension member one by one. It simplifies the configuration process.



Presence

Extension Presence Overview

This topic introduces what is presence and how the presence status can benefit the user's work.

What is Presence

Extension Presence indicates the availability status of an extension. Presence settings are linked to the Call Forwarding rules and Linkus ring strategy. Different call forwarding rules and ring strategy can be set for each presence status.

Yeastar Cloud PBX provides five presence statuses:

- **Available**: The user is online and ready for communication.
- **Away**: The user is currently away from your office.
- **DND**: The user doesn't want to be disturb, and you won't receive any calls.
- **Lunch Break**: The user is currently on lunch break.
- **On a Business Trip**: The user is currently on a business trip.

How does Presence benefit the user's work?

Presence status and information that are displayed on Linkus clients allows the user to see the presence of your colleague and instantly know whether the colleague is available, busy, or away.

Change Presence status to quickly route incoming calls. For example, if the user is at a meeting and do not want to miss calls, set the status to **Away** and forward the call to voicemail. Once the user is ready to receive calls again, switch back to **Available**.

How to change Presence status?

3 ways to change an extension presence.

- On the Linkus client, extension users can change their own presence status.
- On the Extension Web Portal, Linkus users can change their own presence status.
- On the PBX Web Portal, you can change all extensions presence.

Set Call Forwarding Rules & Presence Status

Set call forwarding rules for each presence status. The call forwarding rules allows the user to automatically forward an incoming call to voicemail, another extension, or mobile depending on the extension status.

1. Go to **Settings > PBX > Extensions**, search and find the desired extension, click .
2. Click the **Presence** tab.
3. In the **Presence** drop-down list, select a status to configure.
4. In the **Presence Information** field, enter the a custom status message to display on Linkus.

   The Linkus users can see whether you are available to communicate.

5. Set call forwarding rules for the Presence status.
   a. Select the Call Forwarding conditions:
      - **Always**: All the incoming calls will be forward to the destination.
      - **No Answer**: Only the unanswered calls will be forwarded to the destination.
      - **When Busy**: Only the calls that come in while you are talking on the phone will be forwarded.

   b. Beside the selected forwarding condition, select the forwarding destination.
6. Set the ring strategy for the Presence status.
   - **Ring First**: When a call reaches the extension, which terminal will ring first.
   - **Ring Secondly**: If the incoming call is not answered on the terminals that are selected as **Ring First**, the terminals that are selected as **Ring Secondly** will ring.
7. Click **Save** and **Apply**.

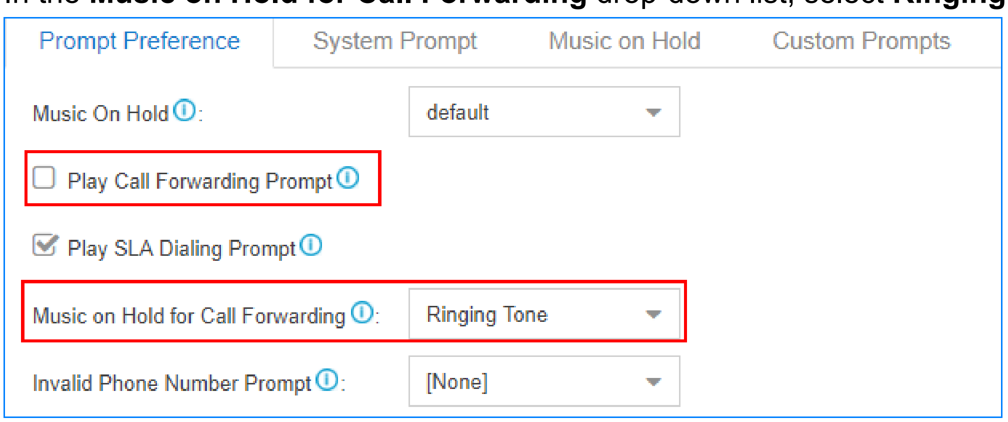
Set Call Forwarding Prompt

By default, when the 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 when the caller is waiting. You can disable the call forwarding prompt and change the MoH music to a normal ring tone. In this way, the caller will not realize that the call is forwarded.

1. Go to **Settings > PBX > Voice Prompts > Prompt Preference**.

2. Unselect the checkbox of **Play Call Forwarding Prompt**.

3. In the **Music on Hold for Call Forwarding** drop-down list, select **Ringing Tone**.



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

Activate/Deactivate Call Forwarding

Extension users can dial the Call Forwarding feature codes on their phones to activate or deactivate Call Forwarding function.

Below are the default call forwarding feature codes and the description of how to use the feature codes.

Code	Action	Example
*71	Activate call forwarding ALWAYS	• Dial *71 to forward all calls to voicemail. • Dial *716000 to forward all calls to extension 6000.
*071	Deactivate call forwarding ALWAYS	• Dial *071 to deactivate call forwarding ALWAYS.
*72	Activate call forwarding WHEN BUSY	• Dial *72 to forward calls (when the user is busy) to voicemail. • Dial *726000 to forward calls (when the user is busy) to extension 6000.
*072	Deactivate call forwarding WHEN BUSY	• Dial *072 to deactivate call forwarding WHEN BUSY.
*73	Activate call forwarding NO ANSWER	• Dial *73 to forward calls (when the user doesn't answer) to voicemail. • Dial *736000 to forward calls ((when the user doesn't answer) to extension 6000.
*073	Deactivate call forwarding NO ANSWER	• Dial *073 to deactivate call forwarding NO ANSWER.

Voicemail

Voicemail Overview

Yeastar Cloud PBX integrates a free voicemail system. Voicemail is a modern kind of answering machine that allows the callers to leave audio messages in case of unavailability.

Enable/Disable Voicemail Function

By default, the voicemail is enabled for all extension users. You can disable the Voicemail function if the user doesn't need it.

1. Go to **Settings > PBX > Extensions**, search and find the desired extension, click .
2. Click the **Presence** tab.
3. Change the Voicemail settings.
    - To enable voicemail, select the checkbox of **Enable Voicemail**.
    - To disable voicemail, unselect the checkbox of **Enable Voicemail**.
4. Click **Save** and **Apply**.

Change Voicemail PIN/Password

Extension users can dial voicemail feature code (default *2) on their phones to access their voicemails. To enhance the extension security, you can change the voicemail PIN on PBX web interface.

1. Go to **Settings > PBX > Extensions**.
2. Search and find the desired extension, click .
3. Click the **Presence** tab.
4. In the **Voicemial Access PIN** field, enter a numeric PIN/password.
5. Click **Save** and **Apply**.

Configure Voicemail to Email

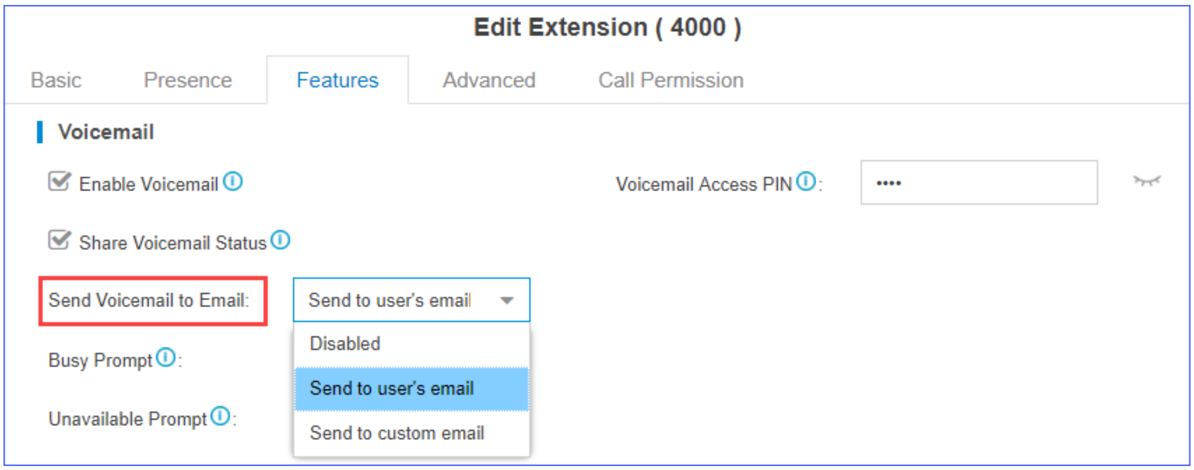
The Voicemail to Email feature of Yeastar Cloud PBX allows extension users to receive voicemail audio files as email attachments and quicken response time when they are out of office.

Enable Voicemail to Email

Voicemail to Email function is disabled by default. If an extension user would like to check voicemail messages via email, you need to enable Voicemail to Email for his/her extension.

Note: To receive voicemail via email successfully, make sure the system email is working.

1. Go to **Settings > PBX > Extensions**, select the desired extension, click .
2. Click the **Features** tab.
3. In the **Send Voicemail to Email** drop-down list, select an email type.



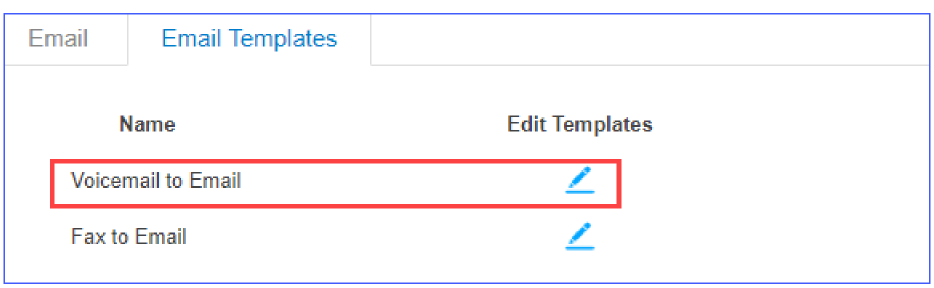
- **Send to user's email**: Send voicemail to the extension user's email address.
- **Send to custom email**: Send voicemail to a custom email address.

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

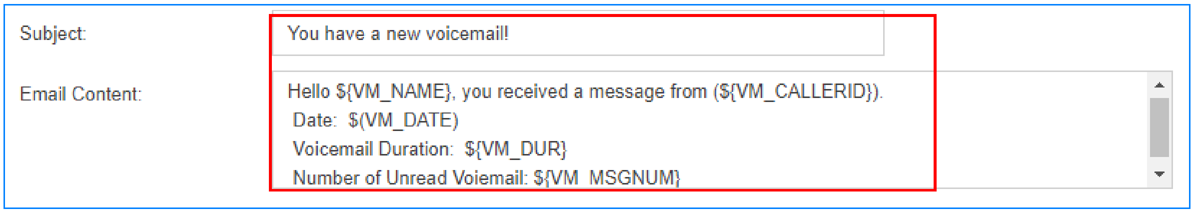
Email template of 'Voicemail to Email'

The PBX has a default email template for **Voicemail to Email**. You can edit the template according to your needs.

1. Go to **Settings > System > Email > Email Templates**, click beside **Voicemail to Email**.



2. Edit the email subject and email contents.



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

Check Voicemail Messages

Extension users have multiple ways to check their voicemail messages.

Check Voicemail on Linkus

Log in Linkus, go to **Me > Voicemail** to check your voicemail.

Check Voicemail on a Phone

- Dial feature code *2 on a phone

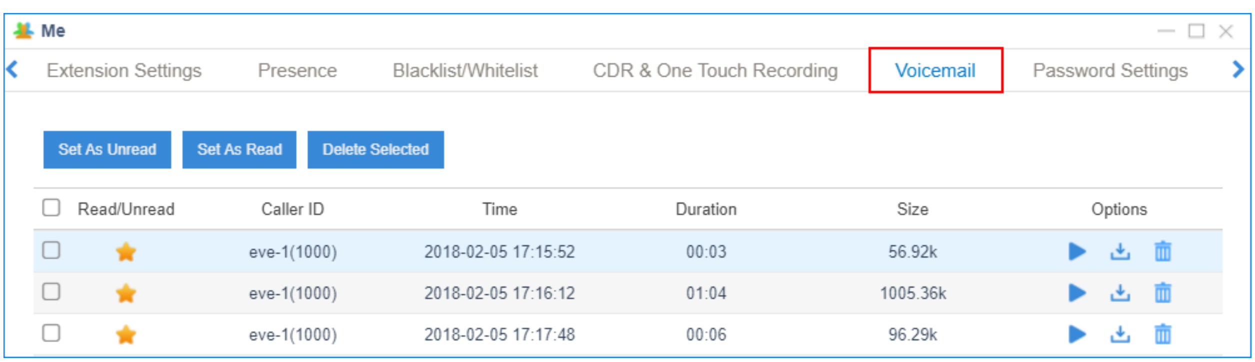
  A user can dial *2 on his own phone to check voicemail.
- Dial feature code *02 on a phone

  A user can dial *02 on other user's phone to enter the voicemail main menu, then enter his/her extension number and voicemail PIN to check voicemail.

Check Voicemail on Web Page

Extension users can log in the PBX web page to check their own voicemails.

- User name: The extension user's email address.
- Password: The extension's **User Password**.



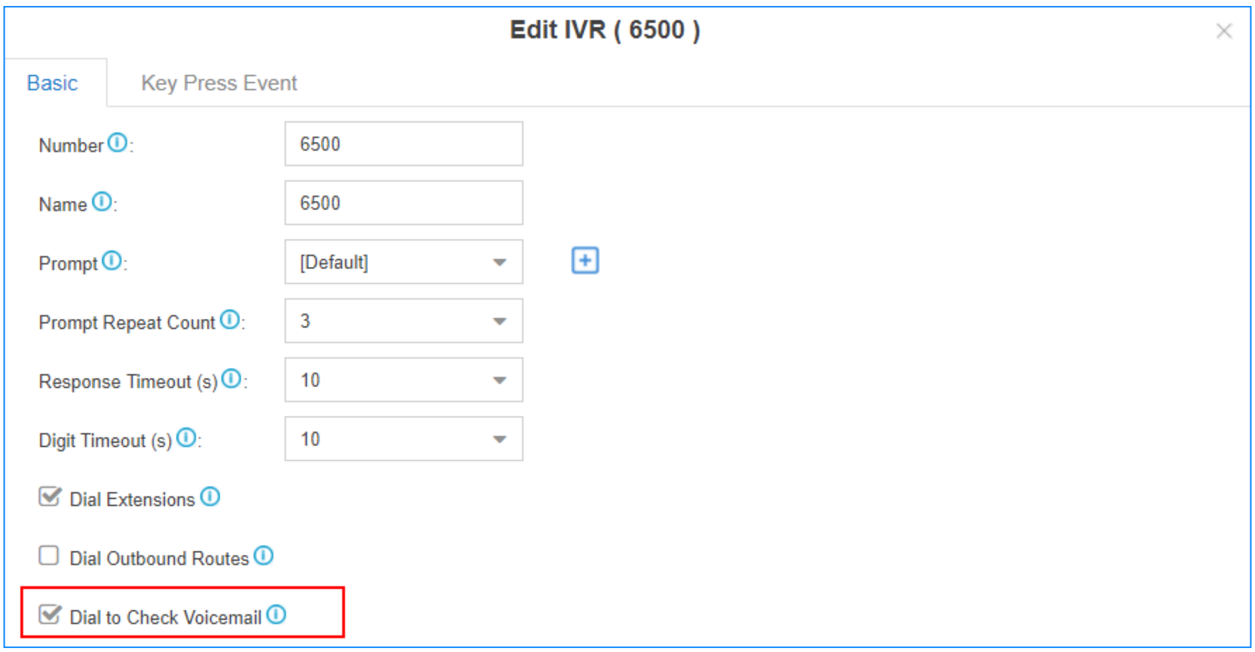
Check Voicemail via Email

If voicemail to email is enabled for an extension user, the user can check voicemails in his/her email box.

Check Voicemail via IVR

If you check the option **Dial to Check Voicemail** for an IVR; users can access the IVR to check their voicemails. This solution is for the users who are outside the office to check their voicemails.

Tip: If the users are using Linkus, they can dial *2 directly to check their voicemails.



Change Voicemail Greetings

You can change the global voicemail greetings for all the extension users or change voicemail greeting for a specific extension.

Components of a Voicemail Greeting

When an extension user is unavailable, the voicemail greeting consists of 3 audio clips: Unavailable Prompt + Voicemail Prompt + "Di".

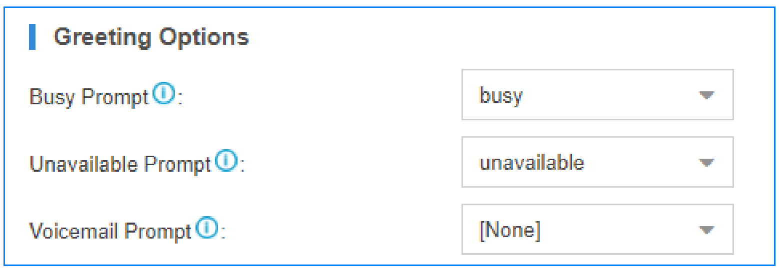
When an extension is busy on a phone, the voicemail greeting consists of 3 audio clips: Busy Prompt + Voicemail Prompt + "Di"

- Default Unavailable Prompt: The person at the extension XXXX is unavailable.
- Default Busy Prompt: The person at the extension XXXX is busy.

- Default Voicemail Prompt: Please leave your message after the tone, when done hang up or press the pound key (#)."

Change global voicemail greetings

1. Prepare your custom prompt files, and upload to the PBX.
2. Go to **Settings > PBX > General > Voicemail > Greeting Options**.
3. Change the global voicemail greetings.
    - **Busy Prompt**: Select the prompt that will be played when the extension is busy.
    - **Unavailable Prompt**: Select the prompt that will be played when the extension is unavailable.
    - **Voicemail Prompt**: Select the prompt that will be played after Busy or Unavailable prompt.



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

Change voicemail greetings for a specific extension

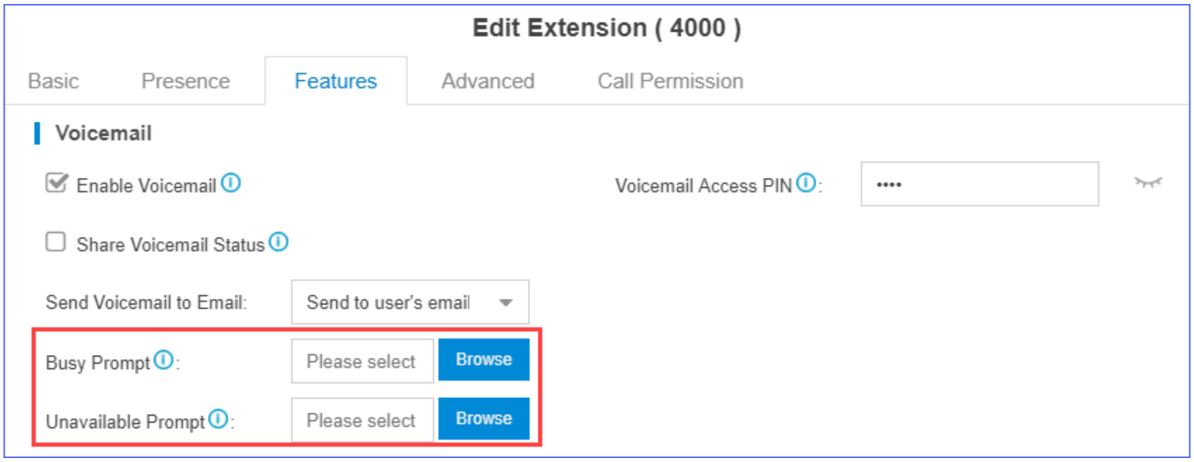
By default, the global busy prompt and global unavailable prompt are applied to all extensions. If an extension user wants to use her/his personal greetings, you can change the prompts for the extension.

Note: The greeting prompt file format should be ".wav", ".WAV" or ".gsm" file.

The file size must not be larger than 8MB.

Supported Format: PCM: 8K, 16bit, 128kbps; A-law(g.711): 8k, 8bit, 64kbps; u-law (g.711): 8k, 8bit, 64kbps; gsm: 6.10, 8k, 13kbps.

1. Go to **Settings > PBX > Extensions**, search and find the desired extension, click .
2. Click the **Features** tab.
3. Click **Browse** to upload a prompt file.



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

Manage Voicemail Messages Centrally

In Yeastar Cloud PBX, you have two options to manage voicemail messages centrally and efficiently: subscribe BLF keys on a phone to monitor multiple extensions' voicemail status and receive multiple extensions' voicemail messages from one mailbox.

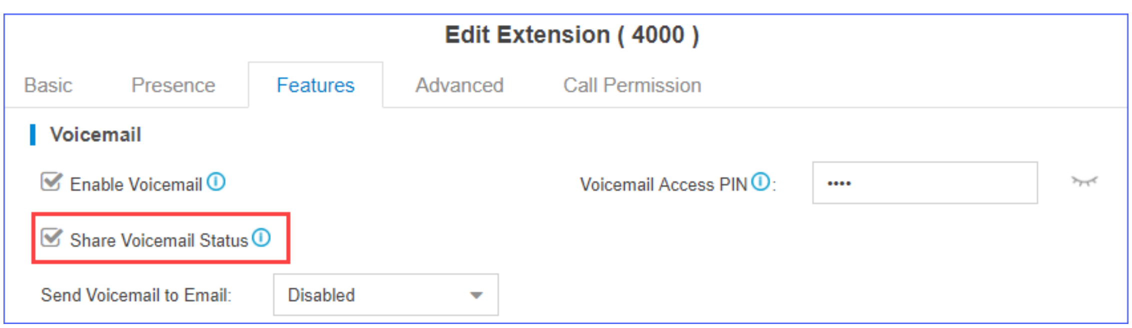
Monitor voicemail status by BLF keys

By default, an extension's voicemail status cannot be monitored by other users. To monitor an extension's voicemail status, you need to enable **Share Voicemail Status** on the extension.

We take Yealink T27G v69.82.0.20 as an example below to introduce how to monitor voicemail status of extension 4000 by extension 1000.

1. **Enable voicemail status sharing feature of extension 4000.**

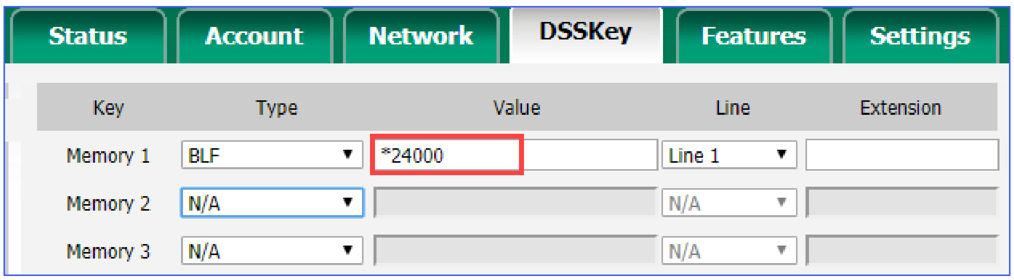
   a. Log in the PBX web interface, go to **Settings > PBX > Extensions**, edit the extension 4000.
   b. On the extension **Features** page, enable **Share Voicemail Status**.



   c. Click **Save** and **Apply**.

2. Set BLF key to monitor the voicemail status.
   a. Log in the IP phone where extension 1000 is registered, go to **Dsskey**.
   b. Set a BLF key to monitor voicemail status of extension 4000.

      - **Type**: Select **BLF**.
      - **Value**: Enter `*2{ext_num}`. In this example, enter *24000.
      - **Line**: Select the line where extension 1000 is registered on.



   c. Click **Confirm**.

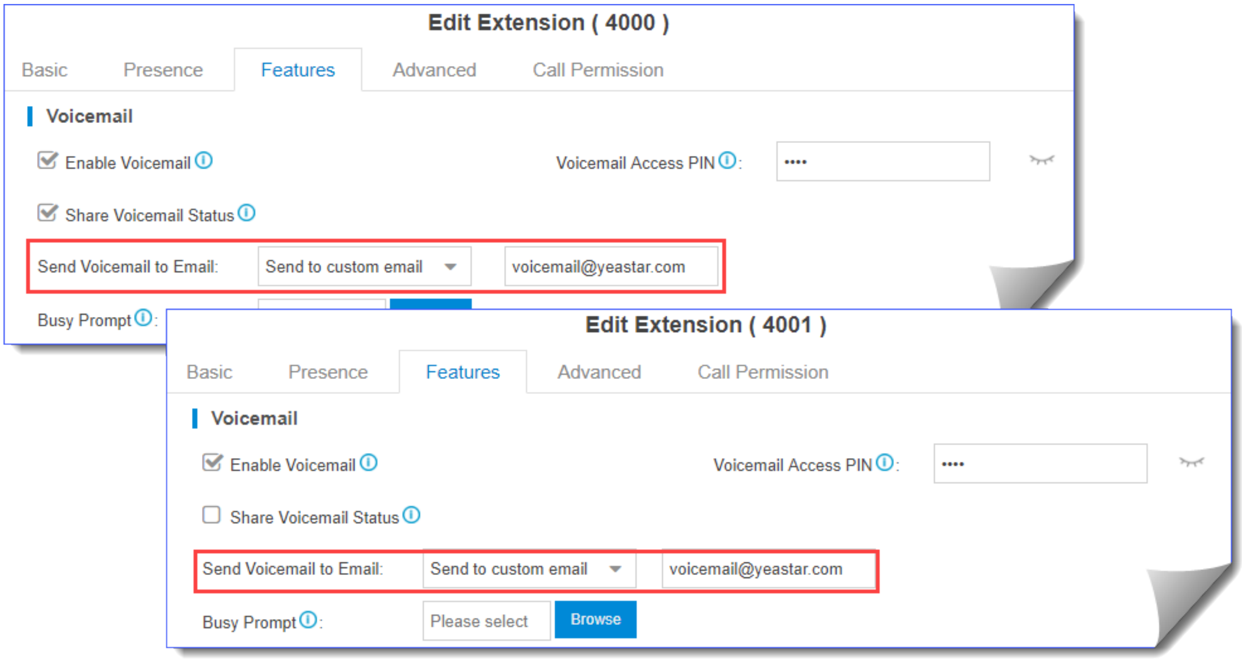
Result:

- **Green BLF LED:** The extension 4000 has NO unread voicemail messages.
- **Red BLF LED:** The extension 4000 has unread voicemail messages.

Receive voicemail from a mailbox

To receive multiple extensions' voicemail messages from one mailbox, you can configure sending voicemail to a same custom email address for these extensions.

For example, to receive multiple extensions' voicemail messages from the mailbox *voicemial@yeastar.com*. Set **Send Voicemail to Email** to the same custom email address *voicemial@yeastar.com* for these extensions.



Global Voicemail Settings

You can change the global voicemail message settings, voicemail playback settings according to your needs.

The global voicemail settings will be applied to all the extensions.

Navigation path: **Settings > PBX > General > Voicemail**.

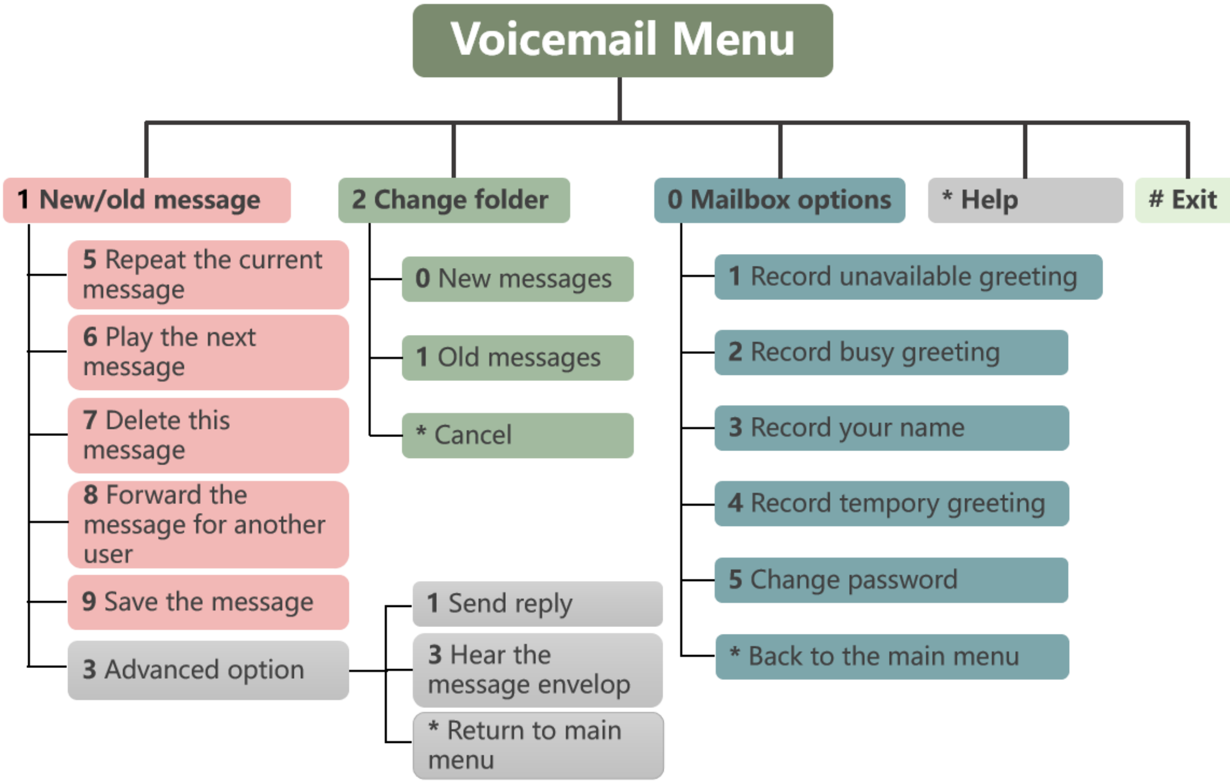
Table 1. Global Voicemail settings

Setting	Description
Message Options	
Max Messages per Folder	Each extension user has a **Read** voicemail folder and an **Unread** folder. You can set the maximum number of messages per folder.
Max Message Time	Set the maximum time of one message.
Min Message Time	Set the minimum time of one message.
Delete Voicemail	This function will work if you enable **Send Voicemail to Email**. If the voicemail is forwarded to the user's email, PBX will delete voicemails from the user's voicemail folder.
Ask Caller to Dial 5	By default, when the caller accesses a user's voicemail, PBX starts to record message automatically. If you want to prompt the caller first, you can enable this option. The caller needs to dial 5 first, then starts to record message.

Setting	Description
Operator Breakout from Voicemail	If enabled, the users can dial 0 to exit the voicemail destination of an IVR.
Greeting Options	
Busy Prompt	Select the greeting that will be played when the extension is busy. **Note:** To use a custom prompt, you need to upload your audio file to the Custom Prompt page first.
Unavailable Prompt	Select the greeting that will be played when the extension is unavailable. **Note:** To use a custom prompt, you need to upload your audio file to the Custom Prompt page first.
Voicemail Prompt	Select the greeting that will be played before the caller leave a message. **Note:** To use a custom prompt, you need to upload your audio file to the Custom Prompt page first.
Playback Options	
Announce Message Caller ID	If enabled, the PBX will announce who left the message.
Announce Message Duration	If enabled, the PBX will announce the message duration.
Announce Message Arrival Time	If enabled, the PBX will announce when the message was received.
Allow Users to Review Messages	If enabled, the users can review their recorded message, and then send the messages.

Voicemail Menu

You can dial *2 on your phone to access the voicemail menu. Below is the detailed voicemail menu.



Call Monitoring

Call Monitoring Overview

Call Monitoring allows authorized users to monitor another extension user's call in real time. The supervisor can dial "feature code" + "extension number" to monitor the extension user's call.

Go to **Settings > PBX > General**, click **Feature Code** tab.

In the **Call Monitor** section, you can enable or disable monitor modes, and modify corresponding feature codes.

Yeastar Cloud PBX supports the following monitor modes:

- **Listen** (Default code: *90)

  Listen mode allows supervisor to listen to a call in real time.

  The supervisor can not talk with the monitored extension users.
- **Whisper** (Default code: *91)

Whisper mode allows supervisor to listen to a call in real time, and talk with the monitored extension user privately.

The other party can not hear the supervisor's voice.

- **Barge-in** (Default code: *92)

  Barge-in mode allows the supervisor to listen to a call in real time and talk with both parties.

Configure Call Monitoring

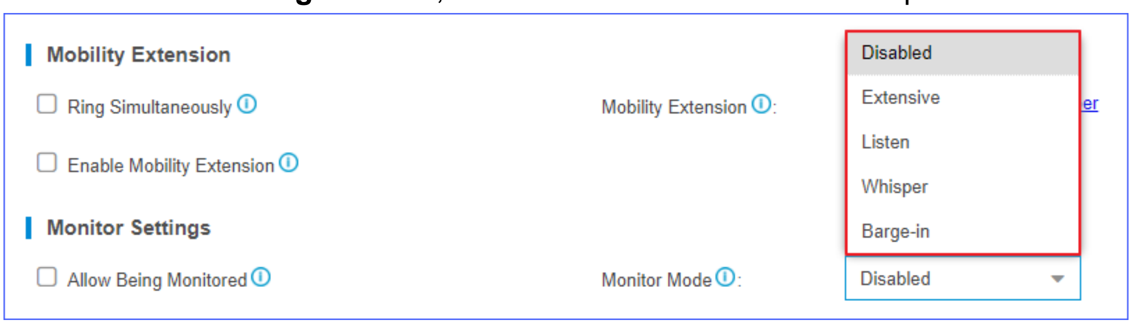
To monitor an extension, you need to set monitor settings for both the supervisors and the monitored extension users.

1. Enable and select a monitor mode for the supervisor.

   a. Go to **Settings > PBX > Extensions**, click beside the desired extension.

   b. On the configuration page, click **Features** tab.

   c. In the **Monitor Settings** section, select a **Monitor Mode** for the supervisor.

   - **Disabled**: Not allowed to monitor other extension users' call.
   - **Extensive**: Use any one of listen, whisper, or barge-in mode to monitor.
   - **Listen**: Listen to a call in real time, but you can not talk with the monitored extension users.
   - **Whisper**: Listen to a call in real time, and talk with the monitored extension users privately.
   - **Barge-in**: Listen to a call in real time and talk with both parties.

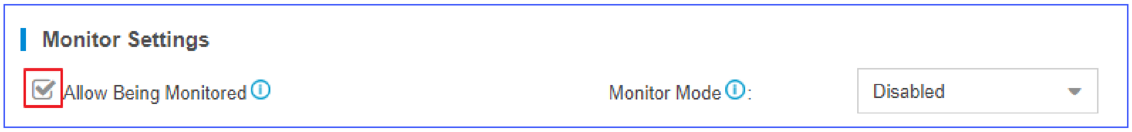
   d. Click **Save** and **Apply**.

2. Set the extension which will be monitored.

   a. Go to **Settings > PBX > Extensions**, click beside the desired extension.

   b. On the configuration page, click **Features**.

c. On the **Monitor Settings** section, select the checkbox of **Allow Being Monitored**.



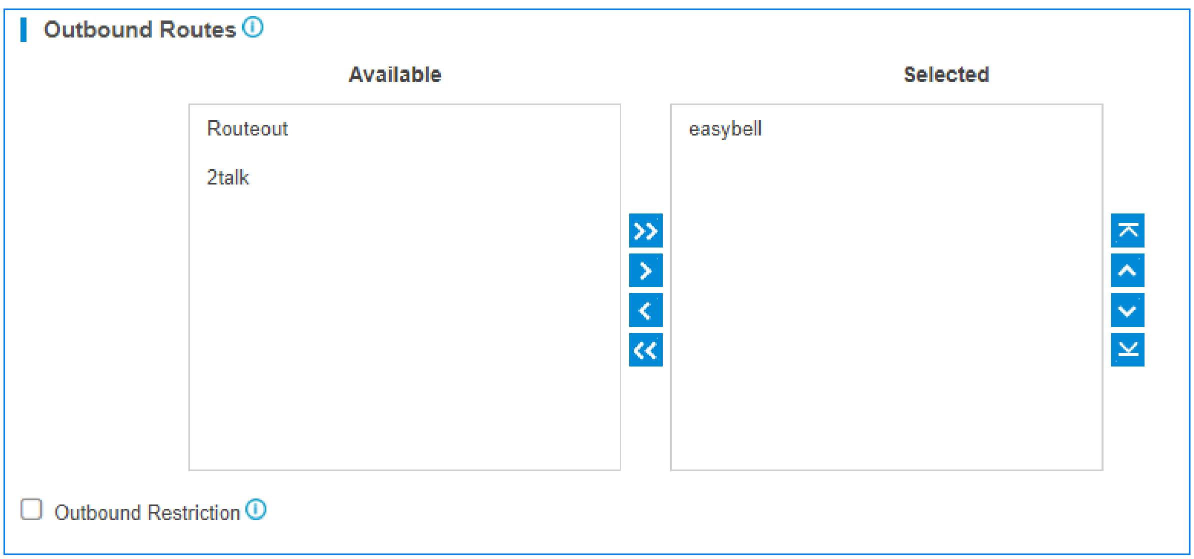
d. Click **Save** and **Apply**.

Call Permission

Set Call Permission of an Extension

On the Extension configuration page, you can set the outbound call permissions for the extension user.

1. Go to **Settings > PBX > Extensions**, click beside the desired extension.
2. On the Extension configuration page, click **Call Permission** tab.
3. Select outbound routes for the extension from **Available** box to **Selected** box.



Outbound Routes Permission

Select outbound routes to the **Selected** box, the extension user will have the permission to make outbound calls through the selected outbound routes.

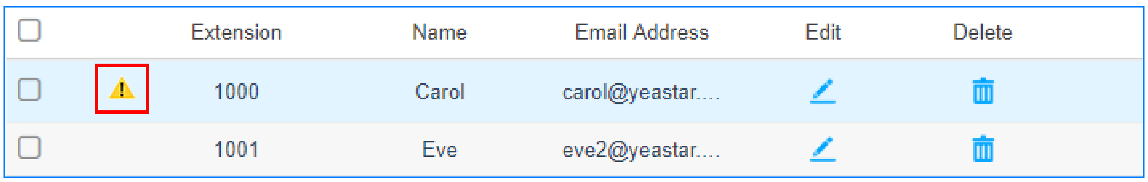
Outbound Restriction

- **Prohibit Outbound Calls**

  Select the **Outbound Restriction** option to prohibit this extension from making outbound calls.

  On the **Extensions** page, the extension will be locked and the extension status will show .

  Note: If the extension user makes outbound calls over the limit of Outbound Restriction rule, the extension will also be locked.



- **Cancel Restriction for Outbound Calls**

  Double click the icon or unselect the checkbox of **Outbound Restriction** to allow this extension to make outbound calls.

Extension Settings

SIP Extension Settings

This reference describes all settings on a SIP extension.

Basic Settings

Navigation path: **Settings > PBX > Extensions**, edit a SIP extension on the **Basic** tab.

General Settings

Settings	Descriptions
Extension	Enter the extension number.
Caller ID	If you set the caller ID number, the called party will see this caller ID number when the extension user makes an outgoing call.
Registration Name	The name used to register a SIP extension.

Settings	Descriptions
Caller ID name	If you set the caller ID name, the called party will see this caller ID name when the extension user makes an outgoing call.
Concurrent Registrations	Yeastar Cloud PBX supports to register one extension number on multiple phones. When a call reaches the extension number, all phones will ring.
Registration Password	The password is used to register a SIP extension. The password is generated randomly by default.

User Information Settings

Settings	Descriptions
Email	Enter the email address. Extension user can reset his/her login password, receive voice mails, faxes, or PBX notifications via this email address.
User Password	The password is used to log in the PBX or log in Linkus mobile client. The password is generated randomly by default.
Prompt Language	The language of voice prompts. The default prompt language is the same as the system language. If the extension user speaks foreign language, you can set a specific system prompt.
Mobile Number	Enter the mobile number. Extension user can receive the PBX notifications or forwarded calls on this mobile number.

Presence Settings

Extension Presence indicates the availability status of a SIP extension. Presence settings are linked to the Call Forwarding rules and Linkus ring strategy. You can set different call forwarding rules and ring strategy for each presence status.

Navigation path: **Settings > PBX > Extensions**, edit a SIP extension under the **Presence** tab.

Presence Settings

Settings	Description
Presence	Set presence status. Yeastar Cloud PBX provides five presence statuses. • **Available**: You are online and ready for communication. • **Away**: You are currently away from your office. • **DND**: You don't want to be disturbed, and you won't receive any calls. • **Lunch Break**: You are currently on lunch break. • **On a Business Trip**:You are currently on a business trip.
Presence Information	Add details about your current status.

Call Forwarding Settings

You can forward calls to a specific destination or a specific extension user to avoid missing calls. Depending on the presence status and your preferences, you can set the PBX to forward calls to voicemail, extension, mobile number, queue, etc.

Settings	Description
Always	Forward all calls to the designated destination.
No Answer	Only forward the unanswered calls to the designated destination.
When Busy	Only forward the calls that come in while you are talking on the phone.

Ring Strategy Settings

You can set ring strategy for the following terminals that the SIP extension registered to.

- Extension
- Linkus Mobile Client
- Linkus Desktop Client

Settings	Description
Ring First	Set which terminal will ring first.
Ring Secondly	Set which terminal will ring secondly.

Features Settings

You can configure voicemail, mobility extension, call monitoring, and other settings under the **Features** tab.

Navigation path: **Settings > PBX > Extensions**, edit a SIP extension under the **Features** tab.

Voicemail Settings

Settings	Description
Enable Voicemail	Enable voicemail feature.
Voicemail Access PIN	Password used to access voicemail.
Share Voicemail Status	Enable this option to share voicemail status of this extension with other extensions.
Send Voicemail to Email	Whether to send voicemail to the designated Email address or not. • **Disabled**: Do not send voicemail to the designated Email address. • **Send to user's mail**: Send voicemail to the email address of the extension user. • **Send to custom mail**: Customize an email address, and the PBX will send the voicemail to the designated Email address.
Busy Prompt	Set the prompt that will be played when the extension user is busy in a call.
Unavailable Prompt	Set the prompt that will be played when the extension user is unavailable.

Mobility Extension

Yeastar Mobility Extension allows you to stay in contact with colleagues and clients using either office phone or mobile phone with the same extension number.

Settings	Description
Ring Simultaneously	Enable this option to allow both extension and associated mobile number ring simultaneously when anyone calls in the extension number.
Enable Mobility Extension	Enable this option to allow your mobile number have the same permission as the office phone when you use associated mobile number to call in the PBX.
Mobility Extension	• **Set Mobile Number**: Set the associated mobile number. • **Prefix**: Set the prefix of the mobile number according to the outbound route.

Monitor Settings

Call Monitoring allows authorized users to monitor another extension user's call in real time.

Settings	Description
Allow Being Monitored	Enable this option to allow anyone to monitor the extension user's ongoing call.
Monitor Mode	Decide how you monitor other extension users' ongoing call. • **Disabled** You can not monitor other extension users' ongoing call. • **Extensive** Use any one of listen, whisper, or barge-in mode to monitor other extension user's ongoing call. • **Listen** Listen to a call in real time, but you can not talk with the monitored extension users. • **Whisper** Listen to a call in real time, and talk with the monitored extension users privately. • **Barge-in** Listen to a call in real time and talk with both parties.

Other Settings

Settings	Description
Ring Timeout (s)	Set the timeout in seconds. Phone will stop ringing after timeout.

Settings	Description
Max Call Duration (s)	Set the maximum call duration in seconds for every call of this extension. **Note:** The precedence of **Max Call Duration(s)** (Global v.s. Extension): • For internal calls: The **Max Call Duration(s)** setting of the caller's extension takes precedence. • For outbound calls: The **Max Call Duration(s)** setting of the caller's extension takes precedence. • For inbound calls: The global **Max Call Duration(s)** setting takes precedence.
Send email notification when extension user password is changed	Enable this option to send email notification when extension user password is changed.

Advanced Settings

The advanced settings of SIP extension require professional knowledge of SIP protocol. Incorrect configurations may cause calling issues. It is wise to retain the default settings provided on the SIP extension page. However, for a few fields, you need to change them to suit your situation.

Navigation path: **Settings > PBX > Extensions**, edit an extension under the **Advanced** tab.

VoIP Settings

Settings	Description
Qualify	Enable this option to send SIP OPTION packet to SIP device to check if the device is up.
Enable SRTP	Enable SRTP for voice encryption.
T.38 Support	Enable or disable T.38 fax for the extension.

Settings	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. • **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.
Transport	Set the transport protocol. • **UDP** • **TCP** • **TLS**

Enable User Agent Registration Authorization

Settings	Description
Enable User Agent Registration Authorization	Whether to restrict user agents from registering to the extension.
User Agent	Enter the name of user agent. If the prefix of the user agent does not match the value, the registration will fail.

IP Restriction

Settings	Description
Enable IP Restriction	This option is used for IP access control. Only the IP address or IP section that matches the settings can register the extension number.

Settings	Description
Permitted IP/Subnet mask	Enter the IP address and subnet mask. • *192.168.5.100/255.255.255.255* In this example, only the device whose IP address is *192.168.5.100* can register the extension number. • *192.168.5.0/255.255.255.0* In this example, only the devices whose IP section is *192.168.5.0* can register the extension number.

Call Permission Settings

You can set the outbound call permissions for the SIP extension.

Navigation path: **Settings > PBX > Extensions**, edit a SIP extension under the **Call Permission** tab.

Settings	Description
Outbound Routes	Set outbound routes for the extension.
Outbound Restriction	Enable this option to prohibit this extension from making outbound calls.

Contacts

Contacts Overview

Yeastar Contacts feature allows you to add external contacts to Company Contacts and share the Company Contacts with your organization. Each extension user has a Personal Contacts to create and manage their personal contacts.

Contacts types

Company Contacts

Company Contacts is a phone book that allows you to store a list of external contacts, such as the company's customers, resellers and partners.

Note: By default, only the PBX administrator can view and manage Company Contacts. To share Company Contacts with extension users, refer to Configure Company Contacts Permissions for Users.

Personal Contacts

Personal Contacts is a phone book for each extension user. Users can store a list of external contacts exclusive to themselves, such as direct customers.

Note: Each user's Personal Contacts is visible only to themselves.

Key features

Sync contacts between Linkus clients and PBX

The contacts information is synced automatically between Linkus clients and PBX.

Users can manage contacts both on Linkus and PBX web page.

Note: Requirements of Linkus clients:

- Linkus Android Client: 2.9.6 or later.
- Linkus iOS Client: 2.9.10 or later.
- Linkus for Mac: Coming soon.
- Linkus for Windows: Coming soon.

For more information of contacts management, see Manage Company Contacts and Manage Personal Contacts.

Import and export contacts

Save time and effort by importing and exporting contacts entries.

For more information, see Manage Company Contacts and Manage Personal Contacts.

Identify incoming calls

The contact's name is displayed for incoming calls to your Linkus, desk phone, or other softphones if the contact's information is saved in Company Contacts or Personal Contacts. By knowing who’s calling, the users can handle the calls efficiently.

For more information, see Identify Callers from Contacts.

Configure Company Contacts permissions for users

Control who can view and manage the Company Contacts.

For more information, see Configure Company Contacts Permissions for Users.

Contacts limits

The following table shows the maximum number of contacts supported on the PBX.

Contacts type	Maximum number
Company contacts (total)	1,000
Personal contacts (per extension)	300

Manage Company Contacts

This topic describes how to add, edit, delete, import, and export company contacts on PBX web page.

Requirements

Only the PBX administrator and the authorized users can manage Company Contacts.

For more information of Company Contacts permissions, see Configure Company Contacts Permissions for Users.

Web operations vs. Linkus operations

The authorized users can manage company contacts on both Web and Linkus, but the operation permissions are different.

For more information of Contacts on Linkus, see Linkus Mobile Help (**Me > Settings > Help & Feedback**).

Operations	Web	Linkus
Add	√	√
Edit	√	√
Delete	√	√
Export	√	×
Import	√	×

Add a company contact

1. Go to **Contacts > Company Contacts**.
2. Click **Add**.

3. Enter the contact information.

   Note: The **First Name**, **Last Name** are required fields, and at least one number is required.

4. Click **Save**.

Edit a company contact

1. Go to **Contacts > Company Contacts**.
2. Select a contact, and click .
3. Edit the contact information.
4. Click **Save**.

Delete company contacts

1. Go to **Contacts > Company Contacts**.
2. To delete a single contact, select the contact and click .
3. To delete multiple contacts, select the checkboxes of the desired contacts, and click **Delete**.

Export company contacts

1. Go to **Contacts > Company Contacts**.
2. Click **Export**.

   All the contacts will be exported to a CSV file.

Import company contacts

Before you begin

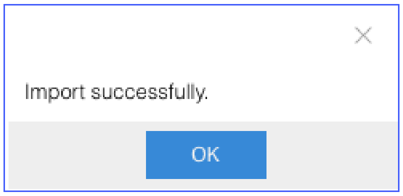
- Prepare a CSV file

  To import contacts, you can export contacts to a CSV file.

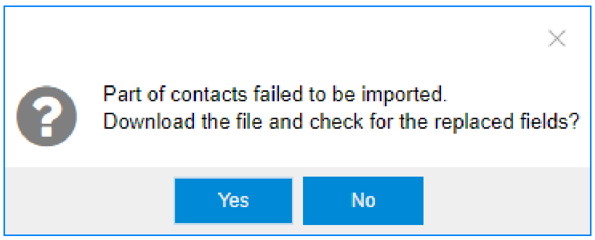
  Use the CSV file as a template, save your data in the same format. For the data requirements in the CSV file, see Import Parameters - Contacts.

1. Go to **Contacts > Company Contacts**.
2. Click **Import**.
3. In the pop-up dialog, click **Browser**, and select your CSV file.
4. Click **Import**.

If the contact data is imported successfully, the web page will display the following confirmation.



If you get an error prompt like the following figure, click **Yes** to check the log and update your data in the CSV file.



Manage Personal Contacts

This topic describes how to add, edit, delete, import, and export personal contacts on PBX web page.

Web operations vs. Linkus operations

Users can manage personal contacts on both Web and Linkus, but the operation permissions are different.

For more information of Contacts on Linkus, see Linkus Mobile Help (**Me > Settings > Help & Feedback**).

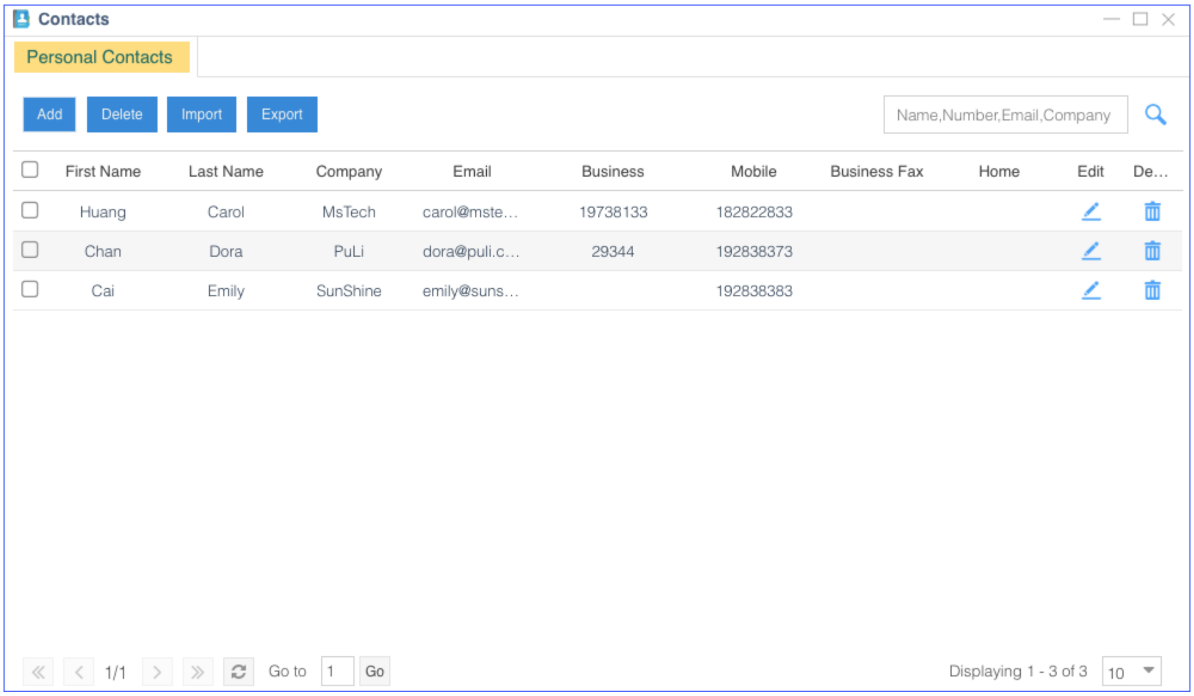
Operations	Web	Linkus
Add	√	√
Edit	√	√
Delete	√	√
Export	√	×
Import	√	×

Access Personal Contacts

Each extension user has a Personal Contacts phone book.

1. Log in PBX web interface using extension email and password.
    - **Username**: Enter extension email.
    - **Password**: Enter the User Password of extension.
2. On the PBX desktop, select **Contacts**.

    The **Personal Contacts** is displayed.



Add a personal contact

1. [Access Personal Contacts on Web.]()
2. On the **Personal Contacts** page, click **Add**.
3. Enter the contact information.

    Note: The **First Name**, **Last Name** are required fields, and at least one number is required.

4. Click **Save**.

Edit a personal contact

1. [Access Personal Contacts on Web.]()
2. Select a contact, and click .
3. Edit the contact information.

4. Click **Save**.

Delete personal contacts

1. Access Personal Contacts on Web.
2. To delete a single contact, select the contact and click .
3. To delete multiple contacts, select the checkboxes of the desired contacts, and click **Delete**.

Export personal contacts

1. Access Personal Contacts on Web.
2. Click **Export**.

   All the contacts will be exported to a CSV file.

Import personal contacts

Before you begin

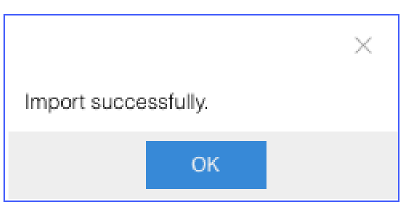
- Prepare a CSV file

  To import contacts, you can export contacts to a CSV file.

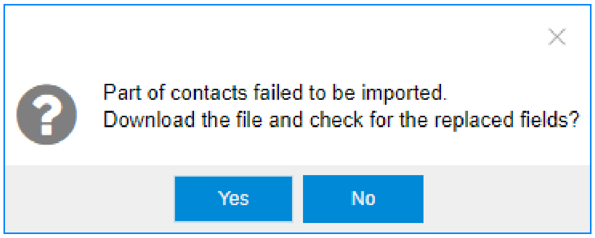
  Use the CSV file as a template, save your data in the same format. For the data requirements in the CSV file, see Import Parameters - Contacts.

1. Access Personal Contacts on Web.
2. Click **Import**.
3. In the pop-up dialog, click **Browser**, and select your CSV file.
4. Click **Import**.

   If the contact data is imported successfully, the web page will display the following confirmation.

   If you get an error prompt like the following figure, click **Yes** to check the log and update your data in the CSV file.



Configure Company Contacts Permissions for Users

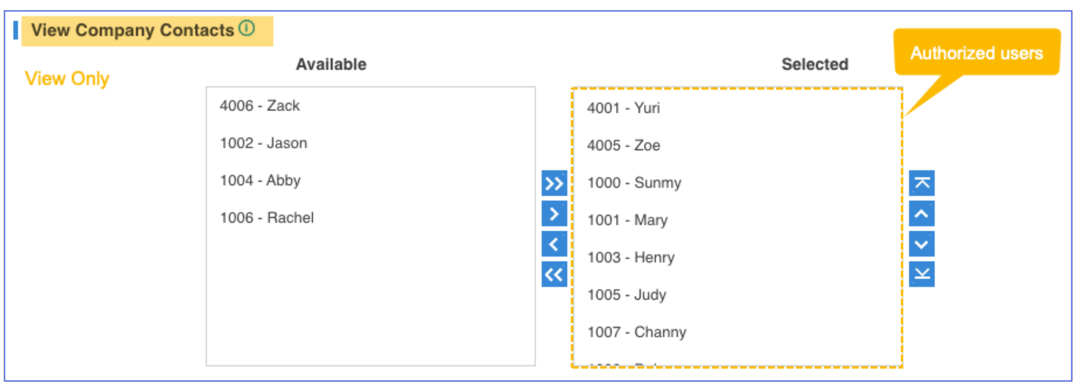
By default, only the PBX administrator can view and manage Company Contacts. To share Company Contacts with your organization, you need to configure Company Contacts permissions for the users in your organization.

Permissions

The PBX provides two permission levels: View and Manage.

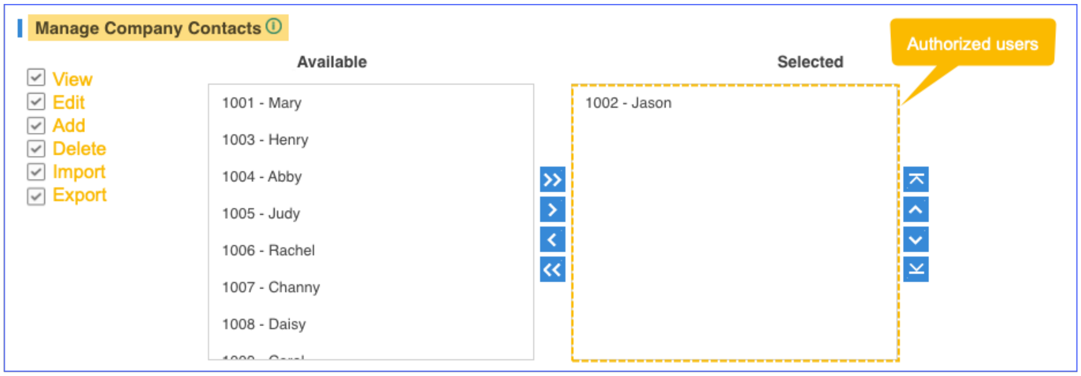
View Company Contacts

The authorized users only have permissions to view the contacts information of the Company Contacts.



Manage Company Contacts

The authorized users have permissions to view, edit, add, delete, import and export the contacts of the Company Contacts.



Configure Company Contacts permissions for users

1. Go to **Contacts > Settings**.
2. To assign View permission to users, configure the section **View Company Contacts**.

   Select the extensions from **Available** box to **Selected** box.
3. To assign Manage permission to users, configure the section **Manage Company Contacts**.

   Select the extensions from **Available** box to **Selected** box.

   Note: Assign the Manage permission carefully to appropriate users. If a user delete contacts accidentally, the contacts would be lost.
4. Click **Save**.

Identify Callers from Contacts

Yeastar Contacts feature allows users to identify incoming callers if the caller information is saved in the Company Contacts or Personal Contacts.

Requirements

Identifying Caller ID is supported on all endpoints, including Linkus, desktop phone, and other softphones.

Identifying callers from Company Contacts

Supported for the authorized users who have permissions to view or manage the Company Contacts.

For more information of the permissions, see Configure Company Contacts Permissions for Users.

Identifying callers from Personal Contacts

Supported for each extension user.

Priority of Caller ID matching

If an incoming number is stored in Company Contacts, Personal Contacts, and mobile phone book at the same time, the priority of Caller ID matching is as follows:

1. Mobile phone book
2. Company Contacts
3. Personal Contacts

Configure Caller ID Match

1. Go to **Contacts > Settings**.
2. Select the checkbox of **Enable Caller ID Match**.
3. Specify to match the exact caller ID or minimum number of caller ID digits.

    - **Exact Match**: Only when the incoming Caller ID matches exactly your existing contact number will the contact name be displayed.
    - **Fuzzy matching**: When the last few digits of the incoming Caller ID matches that of your existing contact number, the contact name will be displayed. The default value is 7.
4. In the **Name Display Format** field, select the contact display order.
    - **First Name Last Name**
    - **Last Name First Name**
5. Click **Save** and **Apply**.

Example

The contact Dora's phone number 12345678 is saved in Company Contacts.

- **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 will not be displayed.
- **Fuzzy matching last 8 digits** is configured:

    - If the incoming caller ID is +012345678, the contact name "Dora" will be displayed.
    - If the incoming caller ID is 62345678, the contact name "Dora" will not be displayed.

Contacts FAQ

- Cannot import my contacts
- Can I set a Contacts sub-administrator?

- Will my personal contacts be lost if I uninstall Linkus client?
- Can the administrator or other users see my personal contacts?
- How do extension users view or manage Company Contacts on PBX web page?
- Does IP phone support Contacts feature?
- Will contacts information be saved when I backup the PBX?
- Can I expand the capacity of Company Contacts?

Cannot import my contacts

1. Check if the contacts limit is reached. See Contacts limits.
2. Check if the imported file meets the format requirements: CSV file encoded in UTF-8 without BOM.

Can I set a Contacts sub-administrator?

Yes.

The PBX administrator can go to **Settings > Permission** to grant **Contacts** permission for the desired user.

If the **Contacts** permission is assigned to an user, the user can do the following operations:

- Manage Company Contacts
- Configure Caller ID Match of Contacts
- Assign Company Contacts permissions to users

Will my personal contacts be lost if I uninstall Linkus client?

The personal contacts won't be lost.

After you create the personal contacts, the contacts is stored in PBX.

Can the administrator or other users see my personal contacts?

No. Personal contacts are visible to the owner.

How do extension users view or manage Company Contacts on PBX web page?

1. Contact administrator to check if you are allowed to view or manage Company Contacts.
2. Log in PBX web interface using extension email and password.
    - **Username**: Enter extension email.
    - **Password**: Enter the User Password of extension.
3. Go to **Contacts > Company Contacts**.

Does IP phone support Contacts feature?

By now, Caller ID match is supported on IP phones, which means that the system can help you identify callers if their information is saved in Contacts.

You can not retrieve and view Contacts information on IP phones at present. But Contacts feature will be supported on IP phones soon, and information can be synchronized among the PBX server, Linkus Clients, and IP phones at that time.

Will contacts information be saved when I backup the PBX?

Yes.

Contacts information is stored on PBX, so it will be automatically saved when you back up the PBX.

Can I expand the capacity of Company Contacts?

No.

Company Contacts is stored on PBX system disk, so you can not expand the capacity by adding extra storage device.

Trunks

Trunk Overview

Making and receiving calls between internal extensions is one thing, but if you want to receive and make calls to the outside world, you need at least a trunk to the outside world.

VoIP Trunks

VoIP Trunks Introduction

VoIP Trunks are phone lines that transmits calls over the Internet. A VoIP provider can assign a local number to one or more cities or countries and route it to the PBX phone system. Usually VoIP trunks are cheaper than traditional PSTN trunks.

VoIP Trunk Types

Yeastar Cloud PBX supports the following VoIP trunk types:

- **VoIP Register Trunk**: Registration based VoIP trunk. VoIP Register Trunk uses the username and password for registration with SIP providers.

- **VoIP Peer Trunk**: Uses the IP address & port or domain of PBX for authentication or connect PBX directly to VoIP provider's dedicated network. Your VoIP provider route incoming and outgoing calls based on the DID number, PBX port or PBX domain, or route calls by a private network.

- **VoIP Account Trunk**

  Account Trunk is designed for connection between Yeastar Cloud PBX and other devices. Yeastar Cloud PBX will act as a VoIP account provider, the other device should register this account to connect to Yeastar Cloud PBX.
- **WebRTC Trunk**

  A WebRTC (Web Real-Time Communication) trunk is used to set up WebRTC Click to Call. After you create a WebRTC trunk on the Yeastar Cloud PBX, a link will be generated automatically.

Create a VoIP Trunk

VoIP Trunk Creation Overview

This topic describes two methods by which to create a VoIP trunk.

VoIP Trunk Creation Methods

Yeastar Cloud PBX supports two methods to create a VoIP trunk.

Create a VoIP Trunk by a Template

Yeastar Cloud PBX supports leading VoIP Service Providers across the globe, you can use the pre-configured VoIP templates included in Yeastar Cloud PBX to set up a VoIP trunk quickly and easily.

Check the tested and supported VoIP providers.

For more information, see Create a VoIP Trunk by a Template.

Create a General VoIP Trunk

If your VoIP provider has not undergone an interoperability test by Yeastar, you can set up a General VoIP trunk.

For more information, see the following topics:

- Create a VoIP Register Trunk - General

- Create a VoIP Peer Trunk - General
- Create a VoIP Account Trunk - General

Create a VoIP Trunk by a Template

If your VoIP trunk provider is tested and supported by Yeastar, you can create a VoIP trunk by a template.

Procedure

1. Go to **Settings > PBX > Trunks**, click **Add**.
2. In the **Name** field, enter a trunk name.
3. From the **Select Country** drop down list, select the country that the VoIP provider operates in.
4. From the **ITSP** drop down list, select the VoIP provider.

   The pre-configured template is applied for the selected VoIP provider.
5. If your trunk is a **Register Trunk**, complete the following configurations:

   a. On the **Basic** page, configure the following settings:

      - **Hostname/IP**: Enter the IP address or the domain of the VoIP provider.
      - **Domain**: Enter the IP address or the domain of the VoIP provider.
      - **User Name**: Enter the username to register to the VoIP provider.
      - **Password**: Enter the password that is associated with the username.
      - **Authentication Name**: Enter the authentication name to register to the VoIP provider.
      - **From User**: Enter the same name as **User Name**.

   b. Configure DID settings for the trunk:

      To add signal DID:

      i. Select **Add Signal DID**.
      ii. Enter the **DID Number** which is provided by the VoIP provider.
      iii. Select the checkbox of **DNIS Name**, enter a DNIS name for the DID number.

         When users call the DID number, the DNIS name will be displayed on ringing phone.

      iv. Click and repeat steps i-ii to add another DID numbers.

      To bulk add DIDs:

      i. Select **Bulk Add DID**.
      ii. Enter the **DID Number Range** which is provided by the VoIP provider, click **Add**.
      iii. Select the checkbox of **DNIS Name**, enter DNIS name for a DID number.

When users call the DID number, the DNIS name will be displayed on ringing phone.

6. If your trunk is a **Peer Trunk**, complete the following configurations:
    - **Hostname/IP**: Enter the IP address or the domain of the VoIP provider.
    - **Domain**: Enter the IP address or the domain of the VoIP provider.
7. Configure other VoIP trunk settings as your need.
8. Click **Save** and **Apply**.

    You can check the trunk status in **PBX Monitor**. If the trunk status shows , the trunk is ready for use.

Create a VoIP Register Trunk - General

If your VoIP provider is not included in the supported VoIP provider list, and you have got a VoIP account with user name and password, you can set up a Register Trunk on Yeastar Cloud PBX.

Assume that you bought a SIP trunk from the VoIP provider, and the trunk information is displayed as below. We will introduce how to set up a Register Trunk according to the trunk information.

Provider address	abc.provider.com
Protocol	SIP
SIP Port	5060
Transport	UDP
Username	254258255
Authenticate name	254258255
Password	05JsOmsIS54SYh
Provided DID numbers	5503301 / 5503302 / 5503303

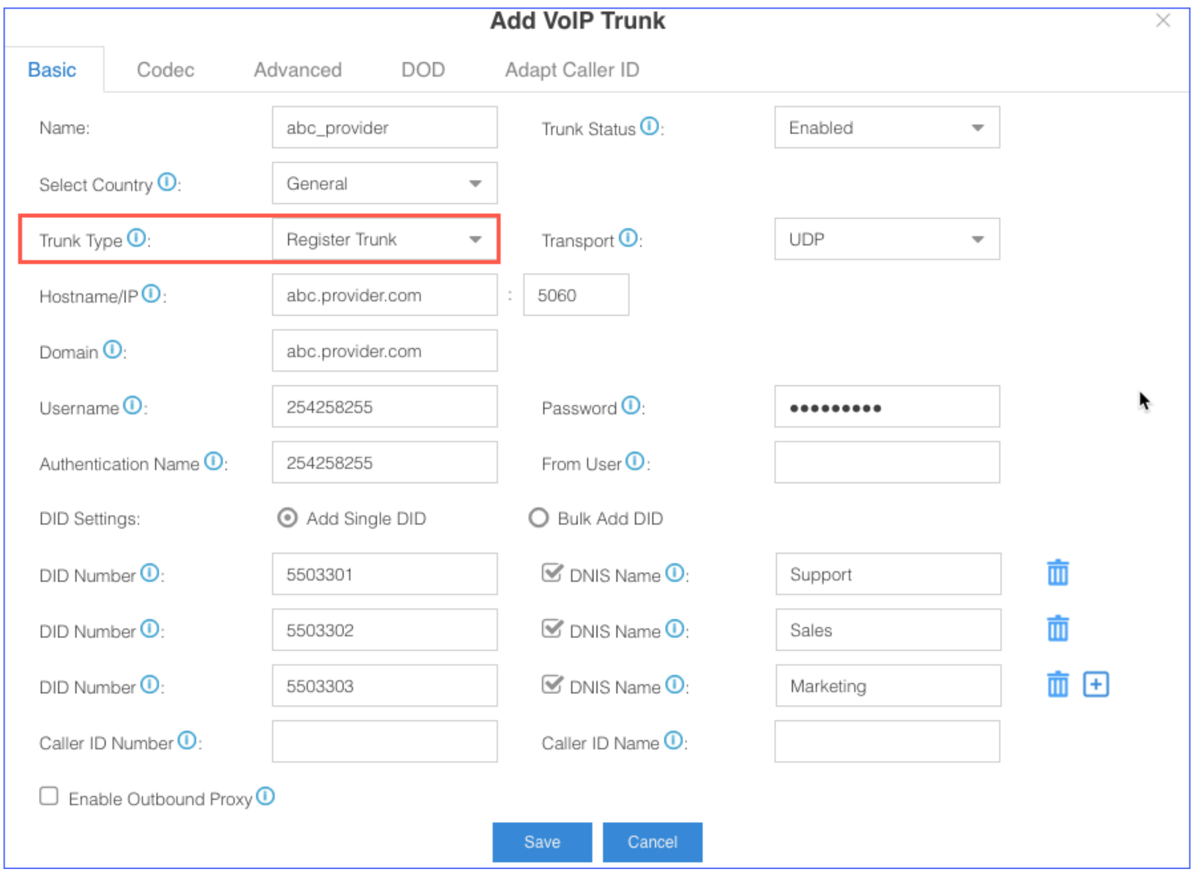
1. Go to **Settings > PBX > Trunks**, click **Add**.
2. In the **Name** field, enter a trunk name.
3. In the **Select Country** drop-down list, select **General**.
4. In the **Trunk Type** drop-down list, select **Register Trunk**.
5. Enter the trunk information that is provided by the VoIP provider:
    - **Hostname/IP**: Enter the IP address or the domain of the VoIP provider (e.g.,*abc.provider.com*).
    - **Domain**: Enter the IP address or the domain of the VoIP provider (e.g., *abc.provider.com*).
    - **User Name**: Enter the username to register to the VoIP provider (e.g., *254258255*).

- **Password**: Enter the password that is associated with the username (e.g., *05JsOmsIS54SYh*).
- **Authenticate**: Enter the authentication name to register to the VoIP provider (e.g., *254258255*).
- **From User**: Enter the same name as **User Name** (e.g., *254258255*).

6. Set DID numbers for the trunk#
   a. Select **Add Single DID**.
   b. Enter the **DID Numbers** which is provided by the VoIP provider.
   c. Select the checkbox of **DNIS Name**, enter a DNIS name for the DID number.

      When users call the DID number, the DNIS name will be displayed on ringing phone.

   d. Click and repeat steps i-ii to add another DID numbers.



7. Configure other VoIP trunk settings as your need.
8. Click **Save** and **Apply**.

   You can check the trunk status in **PBX Monitor**. If the trunk status shows , the trunk is ready for use.

Related tasks
Add an Outbound Route
Add an Inbound Route
Set up DOD Numbers for VoIP Trunk

Create a VoIP Peer Trunk - General

If your VoIP provider is not included in the supported VoIP provider list, and the ITSP only provides an IP address or domain for your purchased VoIP account, you can set up a Peer Trunk on the Yeastar Cloud PBX.

1. Go to **Settings > PBX > Trunks**, click **Add**.
2. In the **Name** field, enter a trunk name.
3. In the **Select Country** drop-down list, select **General**.
4. In the **Trunk Type** drop-down list, select a type of Peer trunk.

   Note: If you don't know which type to select, contact Yeastar support.

Peer Trunk Type	Description
DID-based	If the VoIP provider routes incoming calls and outgoing calls based on the DID number, select DID-based VoIP trunk.
Port-based	If the VoIP provider routes incoming calls and outgoing calls based on the SIP registration port, select Port-based VoIP trunk. **Note:** If this type is selected, a specific SIP registration port will be assigned to the PBX. In this way, the VoIP provide can correctly route the calls.
Domain-based	If the VoIP provider routes incoming calls and outgoing calls based on the PBX domain name, select Domain-based VoIP trunk.
Private Network	If the PBX and the VoIP provider are in the same private network, select Private-Network based VoIP trunk.

5. Enter the trunk information that is provided by the VoIP provider.
   - **Hostname/IP**: Enter the IP address or the domain of the VoIP provider.
   - **Domain**: Enter the IP address or the domain of the VoIP provider.
6. Configure other VoIP trunk settings as your need.
7. Click **Save** and **Apply**.

You can check the trunk status in **PBX Monitor**. If the trunk status shows ✔, the trunk is ready for use.

Related tasks

[Add an Outbound Route]

[Add an Inbound Route]

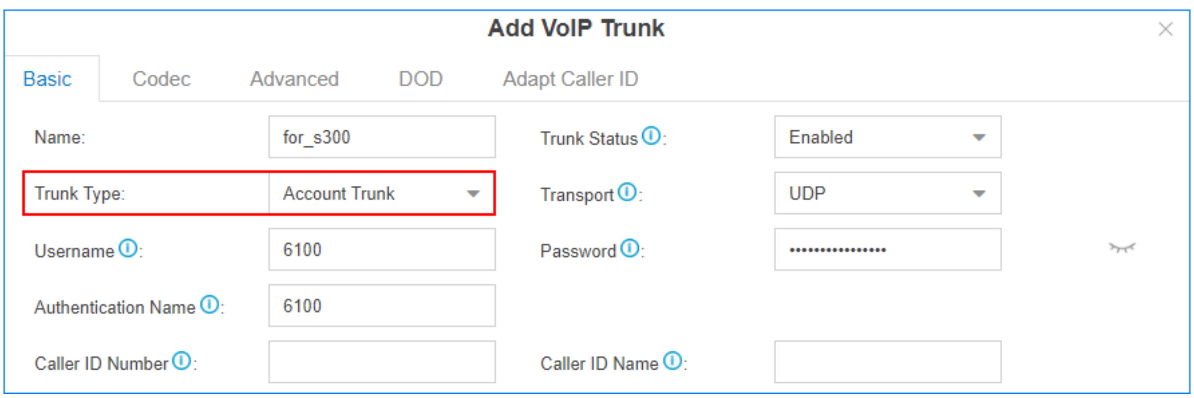
Set up DOD Numbers for VoIP Trunk

Create a VoIP Account Trunk - General

Create a VoIP Account Trunk on the Yeastar Cloud PBX, and provide this account for the other device to register. In this way, Yeastar Cloud PBX and the other device are connected.

1. Go to **Settings > PBX > Trunks**, click **Add**.
2. In the **Name** field, enter a trunk name.
3. In the **Select Country** drop down list, select **General**.
4. In the **Trunk Type** drop-down list, select **Account Trunk**.
5. Enter the account information as your need:
    - **Username**: Use the default or change the number.
    - **Password**: Use the default or change the number.
    - **Authentication Name**: Use the default or change the number.

    Note: The other device should use the provided trunk information to connect to the Yeastar Cloud PBX.

6. Configure other [VoIP trunk settings] as your need.
7. Click **Save** and **Apply**.

    After the Account Trunk is registered on the other device, you can check the trunk status in PBX Monitor. If the trunk status shows , the trunk is ready for use.

Related tasks

[Add an Outbound Route]

[Add an Inbound Route]

Manage VoIP Trunks

Import the VoIP register Trunks

You can create multiple VoIP register trunks by importing a UTF-8 `.csv` file.

For requirements of the import parameters, see Import Parameters - Trunks.

1. Go to **Settings > PBX > Trunks**, click **Import**.
2. Click **Download the Template**, add the VoIP register trunks information in the template file.
3. Click **Browse** to upload the template file, and then click **Import**.

Edit the VoIP Trunk

1. Go to **Settings > PBX > Trunks**.
2. Search and find your VoIP Trunk, click .
3. Click the desired tab to edit the VoIP Trunk Settings as your need.
4. Click **Save** and **Apply**.

Delete the VoIP Trunk

1. Go to **Settings > PBX > Trunks**.
2. Search and find your VoIP Trunk, click .
3. Click **Yes** to confirm the deletion.

VoIP Trunk Settings

When you configure a VoIP trunk, you may need to configure some of the advanced settings. This reference describes all the settings on a VoIP trunk.

Basic Settings

Navigation path: **Settings > PBX > Trunks**, edit a trunk on the **Basic** tab.

Settings	Description
Name	Give this trunk a name to help you identify it.
Trunk Status	Enable or disable the trunk.
Select Country	Select the country that the VoIP provider operates in.
Trunk Type	Select a trunk type.
Transport	Select the transport that is provided by the VoIP provider.

Settings	Description
Hostname/IP	Enter the IP address or the domain of the VoIP provider.
Domain	Enter the IP address or the domain of the VoIP provider.
Username	Enter the username to register to the VoIP provider.
Authentication Name	Enter the authentication name to register to the VoIP provider.
Password	Enter the password that is associated with the username.
From User	Enter a name. All the outgoing calls from this trunk will use this name in From header of the SIP invite package.
DID Number	Direct Inward Dialing number, can be used to distinguish incoming calls.
DNIS Name	Dialed Number Identification Service is a telephony service used to identify which number was dialed. Bind a DNIS name for a DID number, when users call the DID number, the DNIS name will be displayed on ringing phone.
Caller ID Number	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. This feature requires support from the VoIP provider.
Caller ID Name	If you set the caller ID name, when users make outbound calls through this trunk, the called party will see this caller ID name instead of the calling party's name. This feature requires support from the VoIP provider.
Enable Outbound Proxy	Set the outbound proxy if the VoIP provider needs.
Enable SLA	After enabling SLA, users can share this trunk to make outbound calls and receive inbound calls by BLF keys on their phones. In this way, Inbound Route settings and Outbound Route settings for the trunk is invalid.

Advanced Settings

The advanced settings of VoIP trunk requires professional knowledge of SIP protocol. Incorrect configurations may cause calling issues. It is wise to leave the default settings provided on the VoIP trunk page. However, for a few fields, you need to change them to suit your situation.

Navigation path: **Settings > PBX > Trunks**, edit a trunk on the **Advanced** tab.

VoIP Settings

Settings	Description
Qualify	Enable this option to send SIP OPTION packet to SIP device to check if the device is up.
DTMF Mode	Set the default mode for sending DTMF tones. • **RFC4733 (RFC2833)**: DTMF will be carried in the RTP stream in different RTP packets 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.
Enable SRTP	Enable or disable SRTP (encrypted RTP) for the trunk.
Send Privacy ID	Whether to send the Privacy ID in SIP header or not.
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.
User Phone	Whether to add the parameter `user=phone` in the SIP INVITE packet. **Note:** Enable this option if the SIP provider requires.

Inbound Parameters

Settings	Description
Get DID From	Decide from which header field will the trunk retrieve DID header. • **[Follow System]** The trunk will follow the global Get DID From setting. • **TO** • **INVITE** • **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** • **Diversion** • **P-Called-Party-ID** • **P-Preferred-Identity**

Settings	Description
Get Caller ID From	Decide from which header field will the trunk retrieve Caller ID header. • **[Follow System]** The trunk will follow the global Get Caller ID From setting. • **From** • **Contact** • **Remote-Party-ID** • **P Asserted Identify**

Outbound Parameters

Configure SIP parameters for outbound calls.

- **Default**: The same as the value in "From".
- **Trunk Username**: The username you configured for the trunk.
- **Extension Number**: The extension number.
- **DOD Number**: The DOD number that you configured to associate with the extension. If the extension doesn't have an associated DOD number, the **Caller ID Number** of the trunk will be taken instead.
- **From User**: The **From User** value that you configured for the trunk.
- **None**: Do not send the parameter with the SIP INVITE packet.

Settings	Description
Remote Party ID	Select which **Remote Party ID** value should be contained in the SIP INVITE headers when making an outbound call.
P Asserted Identify	Select which **P Asserted Identify** value should be contained in the SIP INVITE headers when making an outbound call.
Diversion	Select which **Diversion** value should be contained in the SIP INVITE headers when making an outbound call.
P-Preferred-Identity	Select which **P-Preferred-Identity** value should be contained in the SIP INVITE headers when making an outbound call.

Transfer Parameters

Configure the SIP parameters for transferred calls.

- **Default**: The same as the value in "From".
- **Trunk Username**: The username you configured for the trunk.
- **Extension Number**: The extension number.
- **DOD Number**: The DOD number that you configured to associate with the extension. If the extension doesn't have an associated DOD number, the **Caller ID Number** of the trunk will be taken instead.

- **The Originator Caller ID**: The Caller ID Number of the first caller in cases that the call is transferred.
- **From User**: The **From User** value that you configured for the trunk.
- **None**: Do not send Remote Party ID with the SIP INVITE packet.

Settings	Description
From	Select which **From** value should be contained in the SIP INVITE headers when the call is transferred.
Diversion	Select which **Diversion**value should be contained in the SIP INVITE headers when the call is transferred.
Remote Party ID	Select which **Remote Party ID** value should be contained in the SIP INVITE headers when the call is transferred.
P Asserted Identify	Select which **P Asserted Identify** value should be contained in the SIP INVITE headers when the call is transferred.
P-Preferred-Identity	Select which **P-Preferred-Identity** value should be contained in the SIP INVITE headers when the call is transferred.

Other Settings

Settings	Description
Maximum Channels	Set the maximum number of concurrent calls on the trunk. **Note:** The value `0` means unlimited.
Realm	SIP Realms, also known as domains within SIP networks. Realm is a component within SIP that is used to authenticate users within the SIP registration process. **Note:** By default, the Realm setting is unnecessary. Contact your service provider if you want to configure Realm.
Inband Progress	This Inband Progress setting applies to the extensions which make calls through this trunk. **Note:** 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.

Codec Settings

Each new created VoIP trunk has a default preferred codec list. However, the default codec list may not match the codecs supported by your VoIP provider. In order to maximize the quality of calls and the amount of bandwidth used for calls, you'll want to choose and configure your preferred codec list to match the settings that your VoIP provider supports.

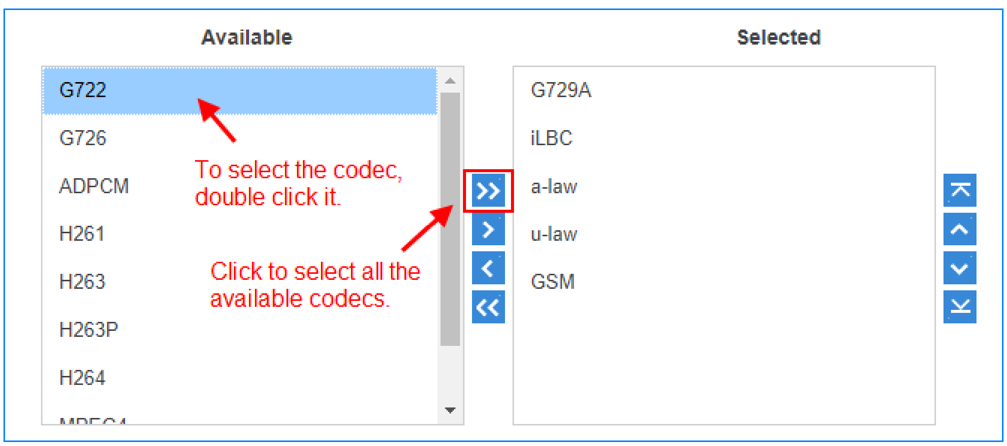
Yeastar Cloud PBX supports the following codecs:

Disabled by default	Enabled by default
GSM, G722, G726, ADPCM, H261, H263, H263P, H264, MPEG4, iLBC	G729, G711 a-law, G711 u-law

Navigation path: **Settings > PBX > Trunks**, edit a trunk on the **Codec** tab.

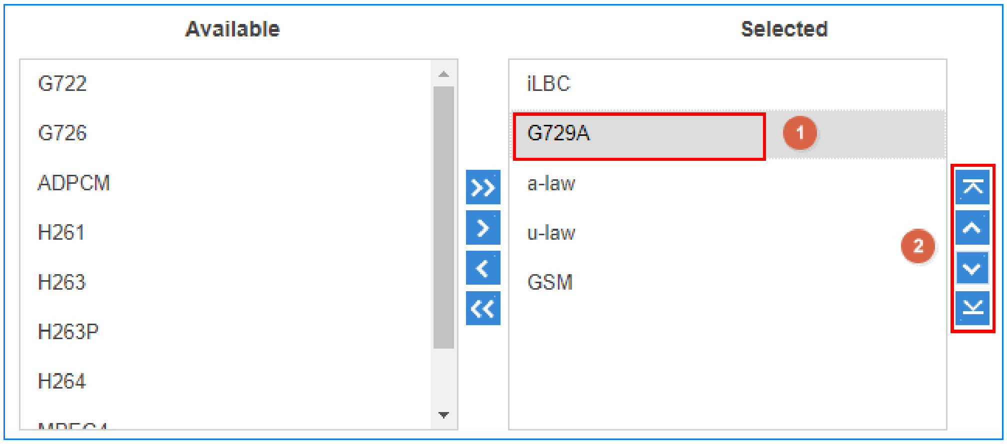
Select Codec

In the **Available** box, double click a codec, the selected codec will appear in the **Selected** box.



Set the Codec Priority

In the **Selected** box, click a codec, and click to change the priority.



Adapt Caller ID

The incoming caller ID that matches the adaptation pattern will be adapted, so that you can press the call record directly on your phone call back a number.

For more information, see Change Inbound Caller ID.

Navigation path: **Settings > PBX > Trunks**, edit a trunk on the **Adapt Caller ID** tab.

Settings	Description
Patterns	The following characters have special meanings: • X matches the numbers 0- 9; • Z matches the numbers 1-9; • N matches the numbers 2- 9; • [12345-9] matches the numbers in the bracket (in this example, 1, 2, 3, 4,5, 6, 7, 8, 9); • Wildcard matches one or more numbers. E.g. "9011." matches anything starting with 9011 (excluding 9011 itself); • Wildcard "!" matches none or more than one numbers. E.g. "9011T matches anything starting with 9011 (including 9011 itself);
Strip	Strip allows you to specify the number of digits that will be stripped from the front of the Caller ID before the call is displayed. For example, if the incoming Caller ID is 05929999999, but you need to dial number 5929999999 to call back, one digit should be stripped.

Settings	Description
Prepend	These digits will be prepended to the Caller ID before the call is displayed. For example, if the incoming caller ID is 5929999999, but you need to dial digit 0 before the number to call back, 0 should be prepended.

WebRTC Trunks

WebRTC Click-to-Call

WebRTC (Web Real-Time Communication) is a collection of communications protocols and application programming interfaces that enable real-time communication over peer-to-peer connections. Yeastar Cloud PBX supports WebRTC Click-to-Call that allows the website visitors calling to a pre-configured destination by clicking a link/button the web page.

Supported Concurrent Calls

A WebRTC trunk supports up to 4 concurrent calls.

Supported Web Browser

We have tested the compatibility for the following browsers that support WebRTC technology.

Note: The failed test is caused by WebRTC not being supported by the web browsers.

Browser	Conditions/Limitations
Google Chrome	Windows Desktop: √ Mac Desktop: √ Android Phone: √ iOS Phone: ×
Firefox	Windows Desktop: √
Opera	Windows Desktop: √ Mac Desktop: √ Android Phone: × iOS Phone: ×
Safari	Safari browser doesn't support WebRTC.

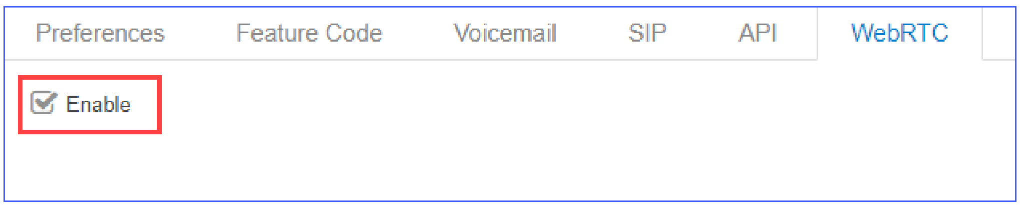
Set up WebRTC Click-to-Call

Create a WebRTC trunk on the PBX, and place the generated link in your website. When a website visitor clicks the link, a WebRTC call will be established between the visitor and the pre-configured destination of the PBX.

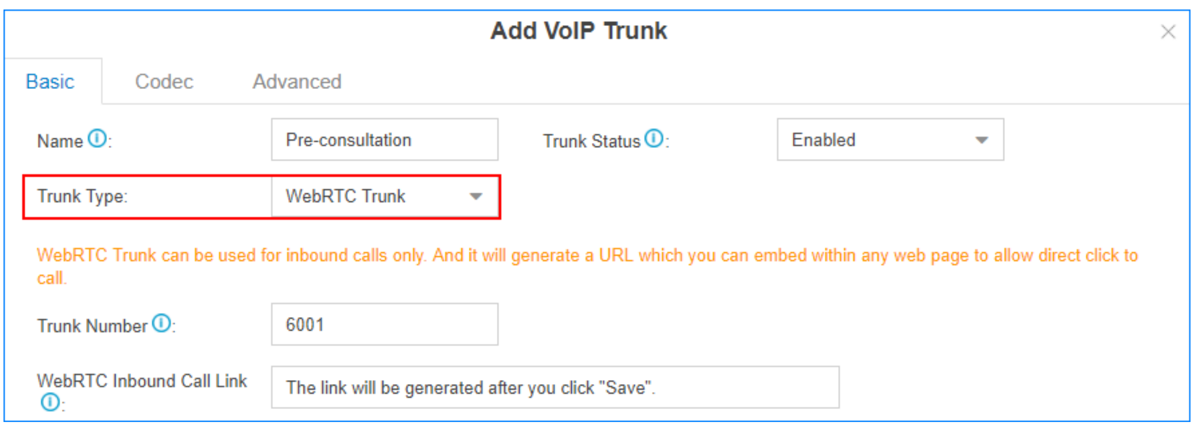
1. Create a WebRTC Trunk

1. Enable WebRTC feature on the PBX.

   Go to **Settings > PBX > General > WebRTC**, check the option **Enable**, click **Save**.



2. Go to **Settings > PBX > Trunks**, click **Add**.
3. In the **Name** field, enter a trunk name.
4. In the **Select Country** drop-down list, select **General**.
5. In the **Trunk Type** drop-down list, select **WebRTC Trunk**.



6. Use the default number or change the **Trunk Number**.

   When a WebRTC call is made through this trunk, the trunk number will be displayed on the ringing endpoint.
7. Click **Save**.

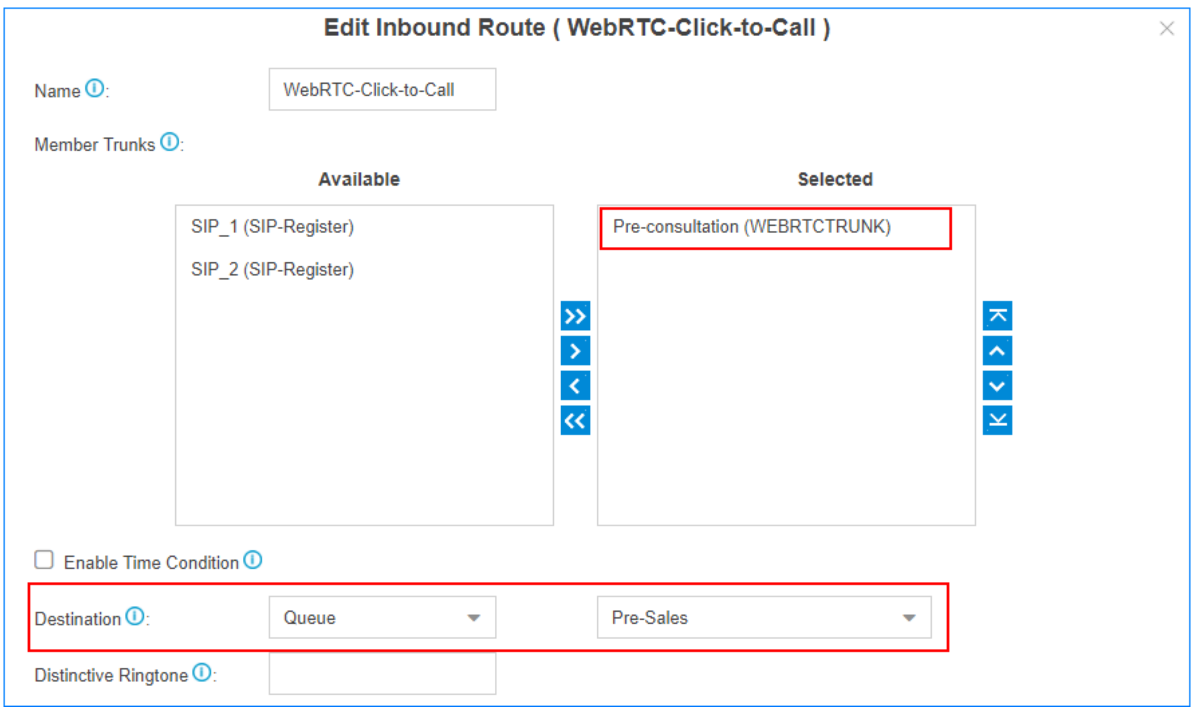
   A link for the WebRTC trunk is generated in **WebRTC Inbound Call Link**. You can place the link on your web page. When your website visitors click the link, they will be connected to the destination of this WebRTC trunk.

8. On the pop-up dialog, click **Copy Now** or **Copy Later**.

2. Set WebRTC Call Destination

Create an inbound route for the WebRTC trunk to route the WebRTC incoming calls. When the website visitors click to call from the web page, the calls will be routed to the configured destination.

1. Go to **Settings > PBX > Call Control > Inbound Routes**, click **Add**.
2. Set WebRTC call destination.
    - **Name**: Enter a route name.
    - **Member Trunks**: Select the WebRTC trunk to the **Selected** box.
    - **Enable Time Condition**: Select the checkbox of **Enable Time Condition**, and configure time conditions to route the incoming calls based on the time conditions.
    - **Destination**: Select the inbound route destination.



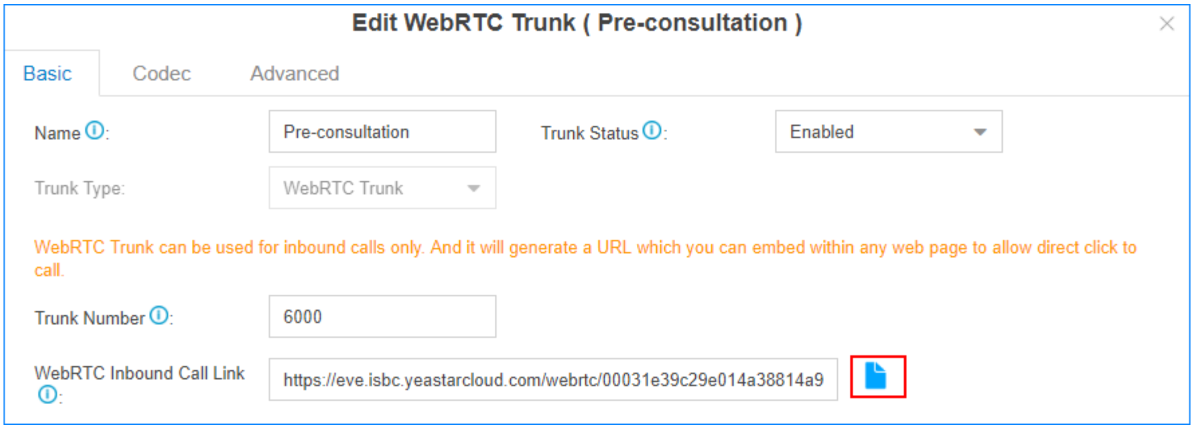
3. Click **Save** and **Apply**.

3. Place WebRTC Link on Your Website

Create an HTML button on your website, and set the button link to WebRTC link that is generated after you creating the WebRTC trunk.

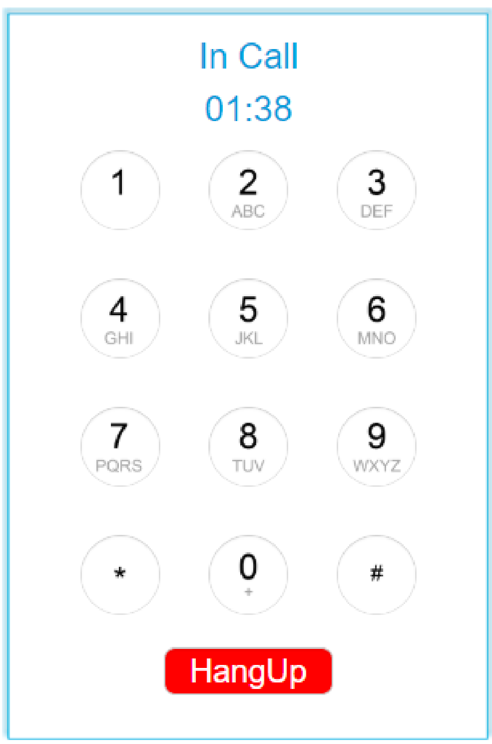
Note: To test the WebRTC Click-to-Call, you can paste the WebRTC link in the web browser directly.

1. On the WebRTC turnk configuration page, click  to copy the WebRTC link.



2. Paste the link your web browser, press **Enter**.

   A dialpad will be displayed on the web page and the call will be connected to your pre-configured destination.



WebRTC Trunk Settings

When you configure a WebRTC Trunk, you may need to configure some of the advanced settings. This reference describes all the settings on a WebRTC Trunk.

Basic Settings

Navigation path: **Settings > PBX > Trunks**, edit WebRTC trunk on the **Basic** tab.

Settings	Description
Name	Enter the trunk name.
Trunk Status	Enable or disable the trunk.
Trunk Type	Select a trunk type.
Trunk Number	Use the default number or change the **Trunk Number**. When a WebRTC call is made through this trunk, the trunk number will be displayed on the ringing endpoint.
WebRTC Inbound Call Link	Place the link on your web page. When your website visitors click the link, they will be connected to the destination of this WebRTC trunk.

Codec Settings

Yeastar Cloud PBX supports a-law and u-law codecs.

Navigation path: **Settings > PBX > Trunks**, edit WebRTC trunk on the **Codec** tab.

Select Codec

In the **Available** box, double click a codec, the selected codec will appear in the **Selected** box.

Set the Codec Priority

In the **Selected** box, click a codec, and click  to change the priority.

Maximum Channel Settings

Navigation path: **Settings > PBX > Trunks**, edit WebRTC trunk on the **Advanced** tab.

Maximum Channels: Defines the maximum number of concurrent calls allowed in this trunk.

Call Control

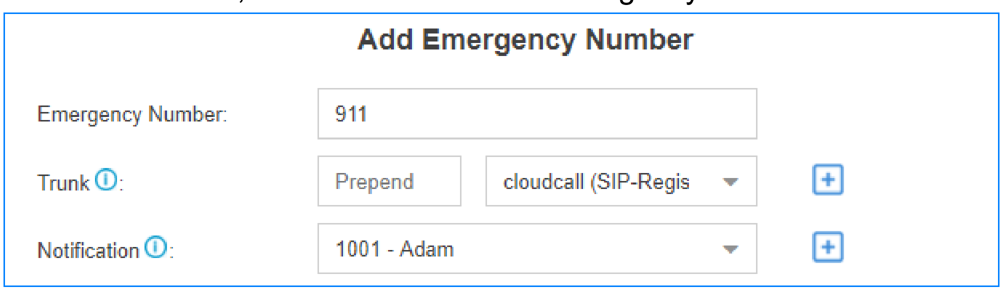
Emergency Numbers

Add an Emergency Number

To ensure that the extension users can make emergency calls at any time, you need to add emergency numbers on Yeastar Cloud PBX. You can also set an alert to notify the emergency contacts that an emergency call has been dialed.

Note: Emergency calls have the highest priority. If the trunk used to make emergency calls is busy, the PBX will terminate the ongoing call, and place the emergency call.

1. Go to **Settings > PBX > Emergency Number**, click **Add**.
2. In the **Emergency Number** field, enter the emergency number.
3. In the **Trunk** field, set the trunk to make emergency calls.



    a. In the drop-down list, select a trunk.
    b. **Optional:** If the selected trunk needs a prepended number before the emergency number, enter a prepended number in the **Prepend** field.
    For example, if your trunk needs a prepended number 0 before the emergency number 911, users should dial 0911 to make the emergency call. To comply with the user's dialing habit, you can set the **Prepend** as `0`. In this way, users can dial 911 as they usually do.
    c. **Optional:** Click to add another trunk.

    Note: If the first trunk cannot work properly, the PBX will use the second trunk to make calls.

4. In the **Notification** field, select the notification contact.

If someone makes emergency calls through the PBX, the contacts will receive notification calls on their extensions.

a. In the drop-down list, select a contact.

b. **Optional:** Click to add another contact.

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

Manage Emergency Numbers

After you add emergency numbers, you can edit or delete them.

Edit an emergency number

1. Go to **Settings > PBX > 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. Go to **Settings > PBX > Emergency Number**, click beside the emergency number that you want to delete.
2. In the pop-up window, click **Yes** to delete the selected emergency number.
3. Click **Apply**.

Time Conditions

Time Conditions Overview

A Time Condition is a time group, which can be applied to outbound routes and inbound routes. You can use Time Condition to control calls based on date and time.

What is a Time Condition used for?

A Time Condition contains a time group.

- **Apply Time Condition to an Inbound Route**

  Time Condition is typically used to control the destination of an inbound call based on the date and time.

  You can select a Time Condition and set a corresponding destination for an inbound route. When a call reaches the PBX, PBX will route the call to the destination when the current system time matches the time defined in the Time Condition.

- **Apply Time Condition to an Outbound Route**

  You can also apply Time Condition to an outbound route to limit when the users can use the outbound route.

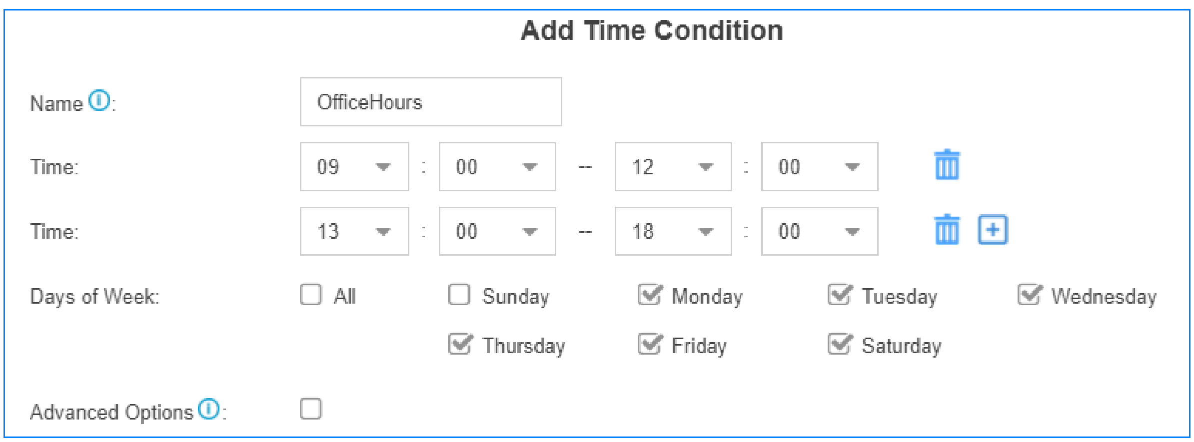
Set Time Conditions

A Time Condition is a time group, which can be applied to outbound routes and inbound routes. This topic describes how to set office hours, non-office hours, and holidays on Yeastar Cloud PBX.

Set office hours

Add a Time Condition according to your office hours. Apply this Time Condition to inbound routes to route incoming calls during office hours to the corresponding destination.

1. Go to **Settings > PBX > Call Control > Time Conditions > Time Conditions**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Time** field, set the time according to your office time.
4. Click [+] to add another time period.
5. In the **Days of Week** field, select your office days.



6. If you want to apply the time period(s) to specific dates, select the checkbox of **Advanced Options**, and set the month and the days of month.

   Note: Advanced Options is disabled by default, which means that the time period(s) will be applied throughout the year.

7. Click **Save** and **Apply**.

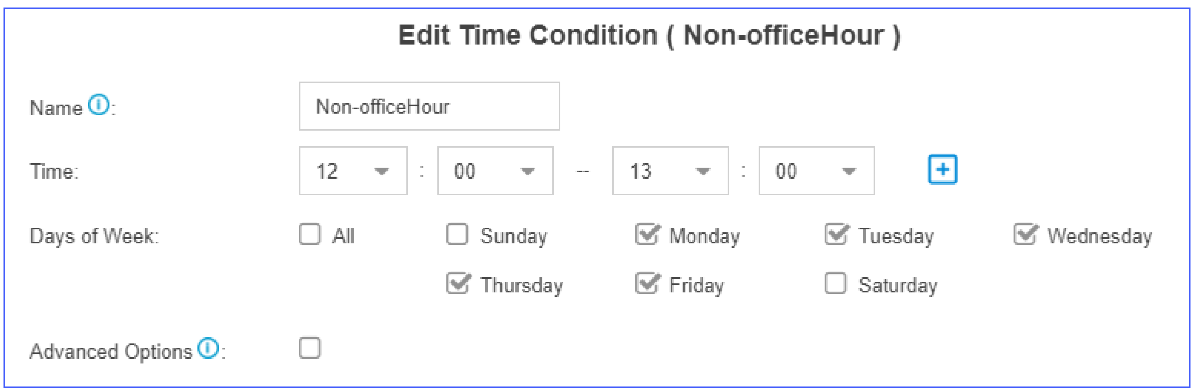
Set non-office hours

PBX has a default Time Condition-**Other Time**. Generally, when you're configuring an inbound route, you can set one destination for office hours, and set the other destination for Other Time.

However, you may need to add another Time Condition to route incoming calls to other destinations due to company's schedule. For example, you want all incoming calls during lunch break to be routed to the receptionist. In this way, employees can enjoy nap time without missing any important calls.

In this case, you can add another Time Condition for non-office hours.

1. Go to **Settings > PBX > Call Control > Time Conditions > Time Conditions**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Time** field, set the time according to your non-office time.
4. Click to add another time period.
5. In the **Days of Week** field, select your office days.



6. If you want to apply the time period(s) to specific dates, select the checkbox of **Advanced Options**, and set the month and the days of month.

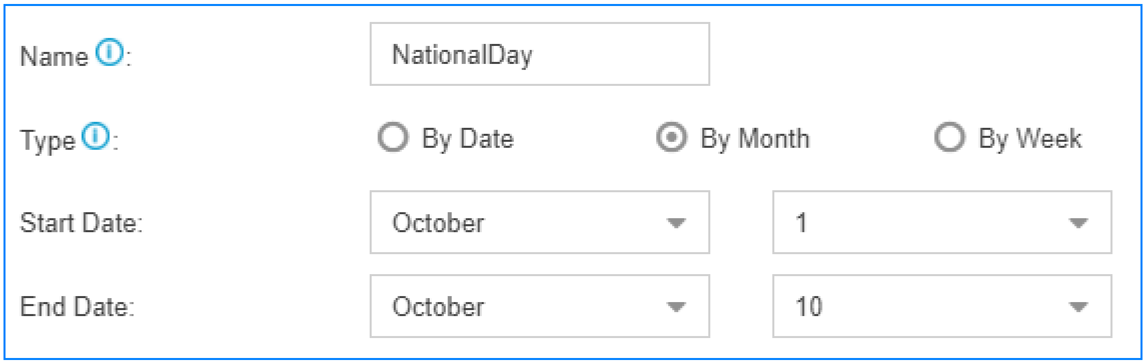
   Note: Advanced Options is disabled by default, which means that the time period(s) will be applied throughout the year.

7. Click **Save** and **Apply**.

Set holidays

You can add a group of holidays and set a Time Condition destination like an IVR for the holidays on your inbound route. When a customer calls to your company during holidays, the PBX will route the call to the IVR and inform your customers that you are on vacation.

1. Go to **Settings > PBX > Call Control > Time Conditions > Holiday**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. In the **Type** field, select a type.



   - **By Date**: If the holiday such as Chinese Spring Festival varies every year, select this type.
   - **By Month**: If the holiday such Chinese National Day always falls on the same calendar date, select this type.
   - **By Week**: If the holiday such as Thanksgiving Day always falls on the same week, select this type.
4. In the **Start Date** field, select the start date of the holiday.
5. In the **End Date** field, select the end date of the holiday.
6. Click **Save** and **Apply**.

Manage Time Conditions

After you create Time Conditions, you can apply them to inbound routes or outbound routes. You can also edit or delete the Time Conditions.

Apply a Time Condition to an Inbound Route

You can apply a Time Condition to an inbound route to route inbound calls to different destinations according to your business hours and schedule.

1. Go to **Settings > PBX > Call Control > Inbound Routes**, click beside the inbound route that you want to edit.
2. On the **Inbound Route** page, select the checkbox of **Enable Time Condition**.
3. Click , and select a Time Condition from the drop-down list.
4. Select destination from the drop-down list.

   Inbound calls will be routed to the pre-configured destination if the date and time of the calls match the time condition.

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

Apply a Time Condition to an Outbound Route

You can apply a Time Condition to an outbound route to limit when the extension users can make outbound calls.

1. Go to **Settings > PBX > Call Control > Outbound Routes**, click beside the outbound route that you want to edit.
2. On the **Outbound Routes** page, select the Time Condition which will be applied to the outbound route.

   Only in this time period can extension users make outbound calls via this outbound route.
3. Click **Save** and **Apply**.

Edit a Time Condition

1. Go to **Settings > PBX > Call Control > Time Conditions**, click beside the Time Condition that you want to edit.
2. Change Time Condition settings according to your needs.
3. Click **Save** and **Apply**.

Delete a Time Condition

After deleting a Time Condition, related configurations of the Time Condition in both inbound routes and outbound routes will be deleted automatically.

1. Go to **Settings > PBX > Call Control > Time Conditions**, click beside the Time Condition that you want to delete.
2. On the pop-up window, click **Yes** and **Apply**.

Time Condition Examples

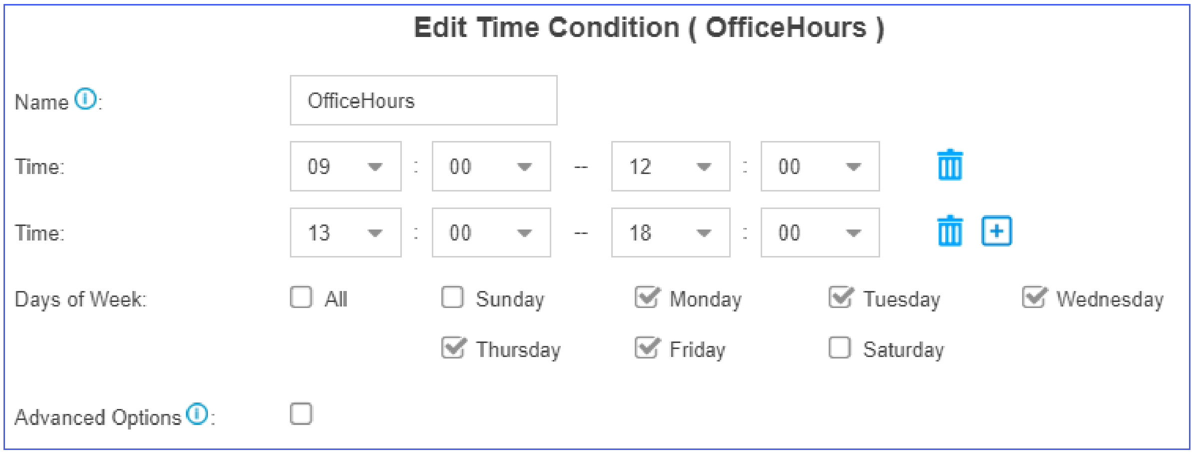
In this topic, we offer you configuration examples of Time Conditions to help you understand how to set office hours, non-office hours, holidays and apply these Time Conditions to inbound routes and outbound routes.

Office hours & non-office hours example

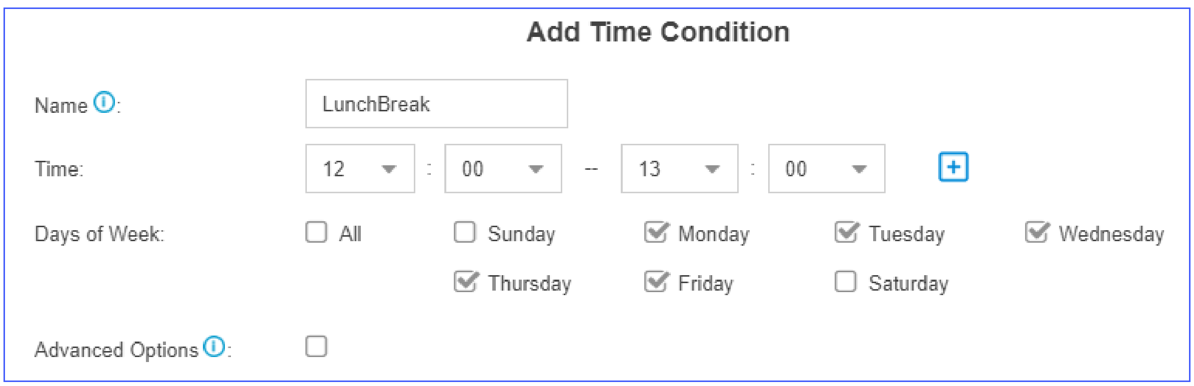
Assume that your office hours are Monday - Friday from 9:00 to 18:00, and the lunch break starts from 12:00 to 13:00.

According to your office hours, you can set two Time Conditions as follows..

- **Office hours**



- **Lunch break**



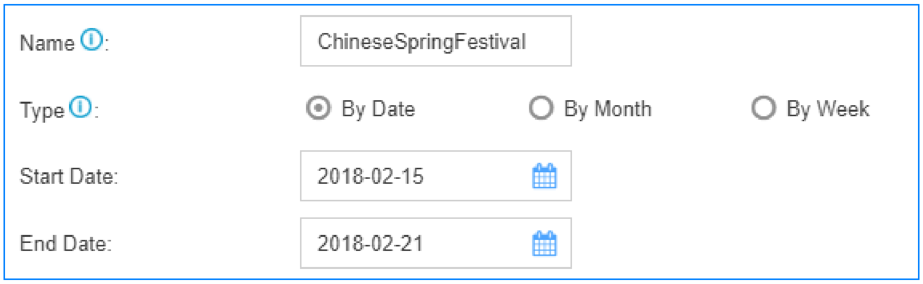
Holiday examples

Yeastar Cloud PBX supports 3 types of holidays.

- **Set a Holiday by Date**

  If date of a holiday varies every year, you can set a holiday by date.

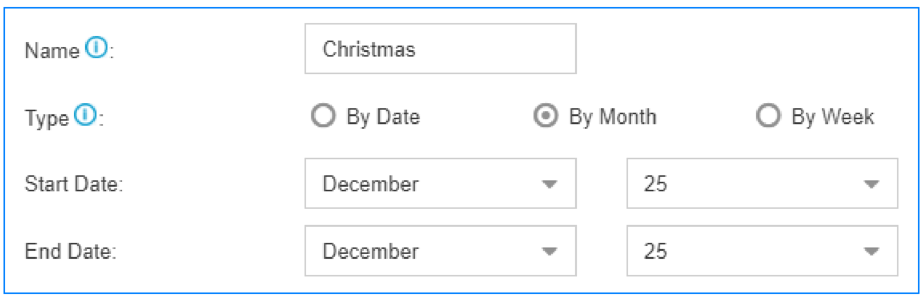
  For example, Chinese Spring Festival falls on February 15th-21st. You can set the holiday as follows.



- **Set a Holiday by Month**

  If a holiday always falls on the same date, you can set a holiday by month.

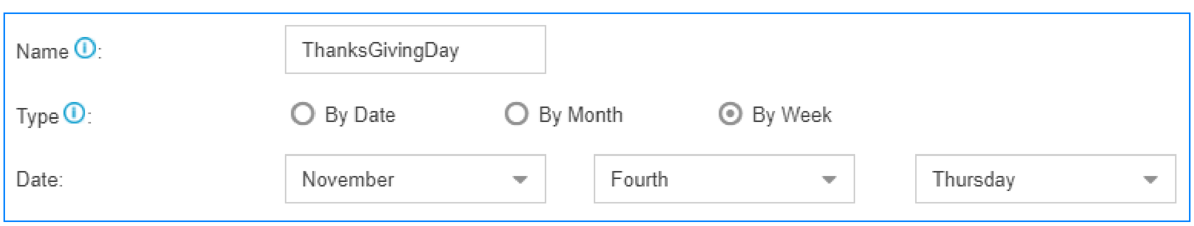
  For example, Christmas falls on December 25th every year. You can set the holiday as follows.



- **Set a Holiday by Week**

  If a holiday always falls on the same week, you can set a holiday by week.

  For example, Thanksgiving Day falls on the 4th week of November. You can set the holiday as follows.



Route inbound calls based on Time Conditions

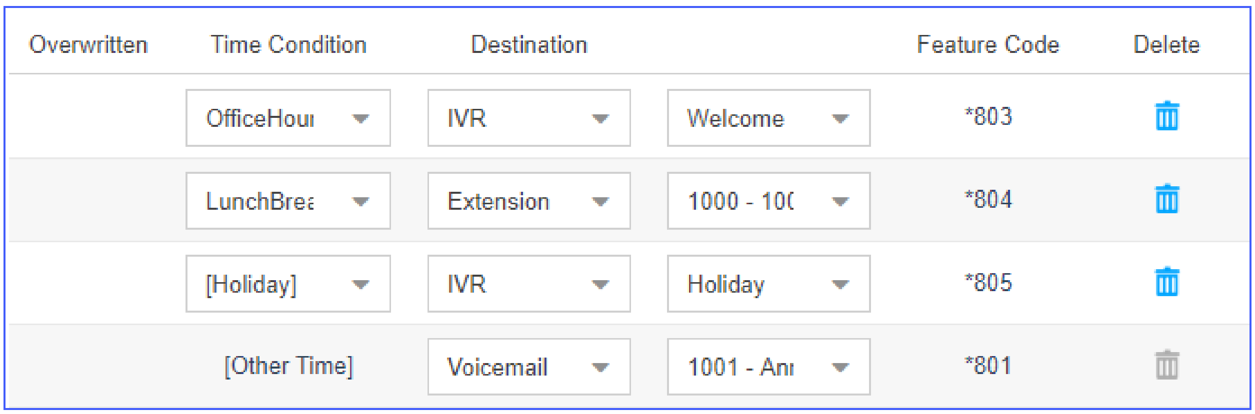
On Inbound Route page, enable **Enable Time Condition**, click to add Time Conditions, and set corresponding destinations.

For example, the following table is a schedule of Time Conditions for a company.

Time Condition	Destination
Office hours	IVR
Lunch break	Extension 1000
Holiday	Holiday IVR
Other time	Voicemail

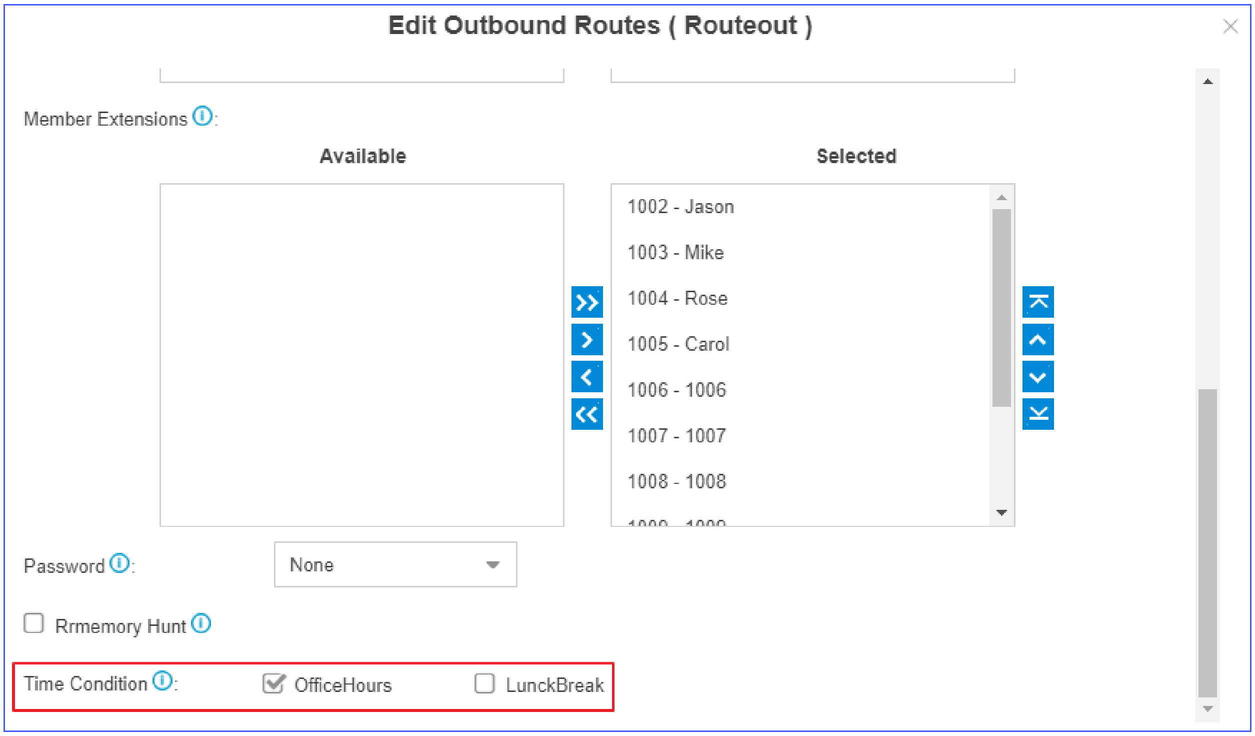
Note: All holidays will be integrated into one **Holiday**, you don't have to select holidays one by one from **Time Condition** on inbound routes.

You can set Time Conditions as follows.



Restrict when to make outbound calls

On Outbound Routes page, select Time Condition, which means that only in this time period can extension users make outbound calls via this outbound route.



Time Condition Override

The Time Condition Override function is used to switch the inbound call routing against the Time Condition. An authorised user can dial Time Condition feature code to override the time condition.

Scenarios

Company A sets day time condition and night time condition in an inbound route with different destinations.

The staffs occasionally leave early or someone needs to enable the night time condition manually. In this scenario, the staffs can dial override feature code to override the time condition.

Time Condition feature code

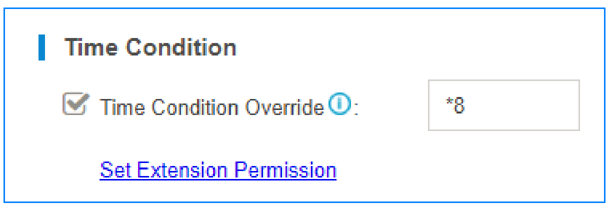
When you enable and add Time Condition on an inbound route, you will see the default generated feature code for the Time Condition. If you want to disable Time Condition Override, dial the Reset feature code *800.

You can go to **Settings > PBX > General > Feature Code > Time Condition** to change the feature code prefix.

Set extension permission to override Time Condition

By default, users have no permission to override Time Condition. You can set which extension users can override Time Condition.

1. Go to **Settings > PBX > General > Feature Code > Time Condition**, click **Set Extension Permission**.



2. Select the desired extensions from **Available** box to **Selected** box.
3. Click **Save** and **Apply**.

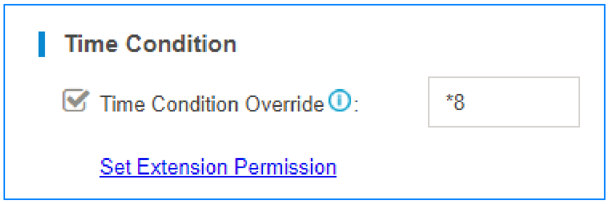
Monitor Time Condition State

You can set a BLF key on your phone to quickly override Time Condition and monitor the Time Condition state.

We take Yealink T53W v95.0.0.0.0.0.1 as an example to explain how to set BLF keys to monitor Time Condition state.

1. Set Time Condition Override permission for the extension that is registered on the IP phone.

    a. Log in PBX interface, go to **Settings > PBX > General > Feature Code > Time Condition**, click **Set Extension Permission**.

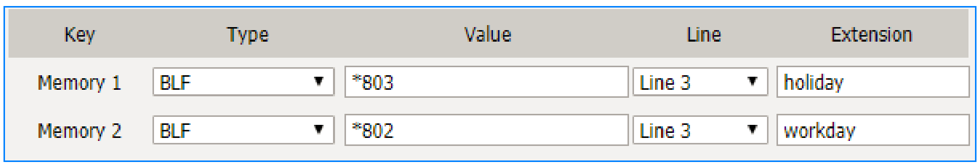


b. Select the desired extension from **Available** box to **Selected** box.

c. Click **Save** and **Apply**.

2. Set BLF keys on the phone where the extension is registered.

a. Log in the phone web interface, go to **DSS Key > Memory Key**.



Key	Type	Value	Line	Extension
Memory 1	BLF	*803	Line 3	holiday
Memory 2	BLF	*802	Line 3	workday

b. Set Key **Type** as **BLF**.

c. Set Key **Value** as feature code of Time Condition.

d. Select the **Line** as the extension registered line.

e. **Optional:** In the **Extension** field, enter a description of the key.

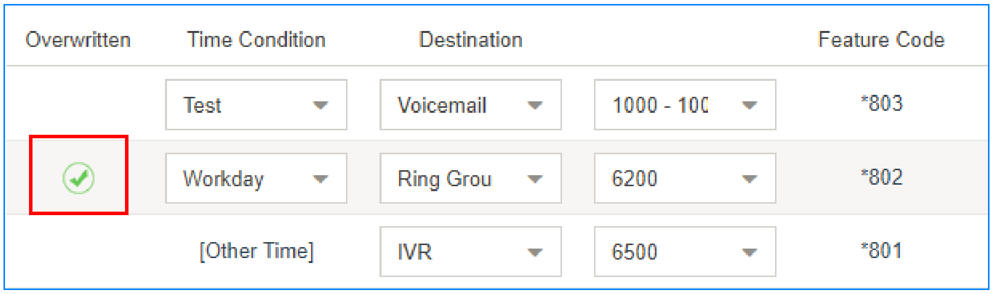
f. Click **Confirm**.

The BLF LED will show the Time Condition state.

- Red: The PBX is using this Time Condition; inbound calls go to the destination of the Time Condition.
- Green: This Time Condition is not in use.

3. Press a BLF key to override Time Condition, the BLF LED turns to red.

You can also log in the PBX web interface, and check the Time Condition state on configuration page of Inbound Routes. If the state shows , it indicates that the PBX is using the Time Condition, and route all incoming calls to destination of the Time Condition.



Inbound Routes

Inbound Route Overview

An inbound route is used to tell the PBX where to route inbound calls based on the caller's phone number or the DID number. Inbound routes are often used in conjunction with time conditions and an IVR.

DID routing & Caller ID routing

Yeastar Cloud PBX allows two specific types of inbound routing: DID Routing and Caller ID Routing. You can set both DID routing and Caller ID routing for an inbound route, or set one of the routing types.

If you don't specify DID numbers and Caller ID numbers on the inbound route, the inbound route will match and route all inbound calls to a pre-configured internal destination on the PBX.

Inbound routes can send inbound calls to destinations as follows:

- Hang up

- Extension
- Extension Range
- Voicemail
- IVR
- Ring Group
- Queue
- Conference
- DISA
- Callback
- Outbound Route
- Fax to Email

Add an Inbound Route

To receive external calls on Yeastar Cloud PBX, you need to set up at least one inbound route.

The PBX has a default inbound route. When users call to the selected trunk, the PBX will route the call to an IVR. You can delete the default inbound route, then add a new one to configure settings according to your needs.

1. Go to **Settings > PBX > Call Control > Inbound Routes**, click **Add**.
2. In the **Name** field, enter a name to help you identify it.
3. **Optional:** In the **DID Pattern** field, enter a DID number or a DID pattern if you want to route inbound calls based on DID numbers.
   The PBX will route the call only when the caller dials the matched numbers.

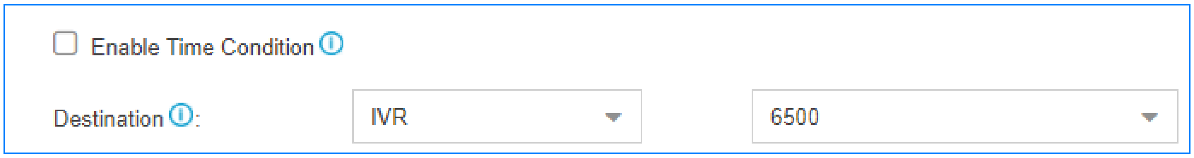
   Note: Leave this blank to match calls with any or no DID info.

4. **Optional:** In the **Caller ID Pattern** field, enter a Caller ID or a Caller ID pattern if you want to route inbound calls based on Caller IDs.
   The PBX will route the call only when the caller ID number matches the **Caller ID Pattern**.

   Note: Leave this blank to match calls with any or no caller ID info.

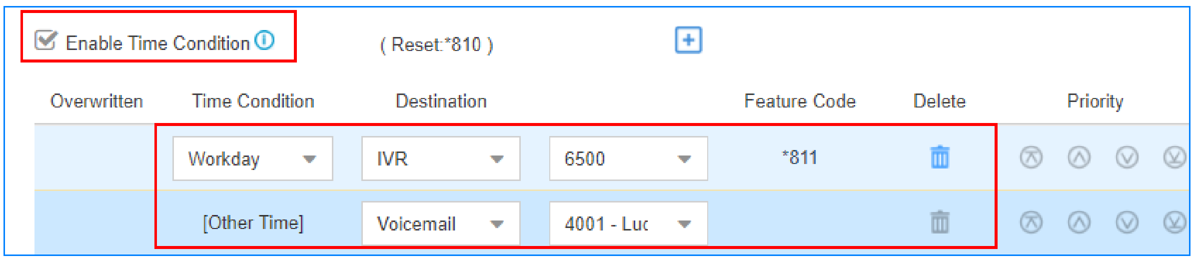
5. In the **Member Trunks** field, select the desired trunk from **Available** box to the **Selected** box.
   The PBX will route the inbound call when the caller calls the number of the selected trunk.

6. If you allow the inbound calls to be routed to a desired destination without time limit, configure the following settings:



   a. Uncheck the checkbox of **Enable Time Condition**.
   b. Select the **Destination**.

7. If you allow the inbound calls to be routed to different destinations based on time condition, configure the following settings:



   a. Select the checkbox of **Enable Time Condition**.

   b. Click , select a Time Condition and the destination.

      If an inbound call reaches the PBX during the time period, PBX will route the call to the selected destination.

   c. **Optional:** Click to set another time condition and destination.

   d. Set the destination for **Other Time**.

      If an inbound call reaches the PBX beyond the time periods that are defined in the above Time Conditions, PBX will route the call to the selected destination.

8. **Optional:** In the **Distinctive Ringtone** field, enter the ringtone name. Distinctive Ringtone helps users recognize where the call is from.

   Note: **Distinctive Ringtone** feature needs support from the IP phones.

   For example, the IP phone has a ringtone called "Family". You can enter "Family" in the **Distinctive Ringtone** field. When a call reaches the IP phone through this inbound route, the IP phone plays the "Family" ringtone.

9. **Optional:** Select the checkbox of **Enable Fax Detection**. PBX will send the fax to **Fax Destination** if a fax tone is detected.
    - **Extension**: PBX will send the fax to **Fax Destination** if a fax tone is detected.
    - **Fax to Email**: PBX will send the fax as an attachment to the specified email address. An email address can be associated with extensions or be customized address.

      Note: If you want to send fax to email, make sure system email is configured correctly.

10. Click **Save** and **Apply**.

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 the selected trunk, the PBX will route the call through the inbound route with higher priority. You can adjust the priority of inbound routes according to your needs.

1. Go to **Settings > PBX > Call Control > Inbound Routes**.
2. Click to adjust the priority of your inbound routes.

Edit an inbound route

1. Go to **Settings > PBX > Call Control > Inbound Routes**.
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. Go to **Settings > PBX > Call Control > Inbound Routes**.
2. Click beside the inbound route that you want to delete.
3. On the pop-up window, click **Yes** and **Apply**.

Import Inbound Routes

You can import inbound routes to quickly set up inbound routing on Yeastar Cloud PBX.

1. Go to **Settings > PBX > Call Control > Inbound Routes**, click **Import**.
2. Click **Download the Template**, add the inbound routes information in the template file.

Note:
- The imported file should be a UTF-8 `.csv` file.
- For requirements of the import parameters, refer to Import Parameters - Inbound Routes.

3. Click **Browse** to upload the template file.
4. Click **Import**.

Change Inbound Caller ID

By default, the Inbound caller ID on Yeastar Cloud PBX displays the caller's phone number, you can change the inbound caller ID with Adapt Caller ID feature.

Adapt Caller ID feature is supported on each trunk. Go to **Settings > PBX > Trunks**, click **Adapt Caller ID** tab on the trunk edit page to configure the settings.

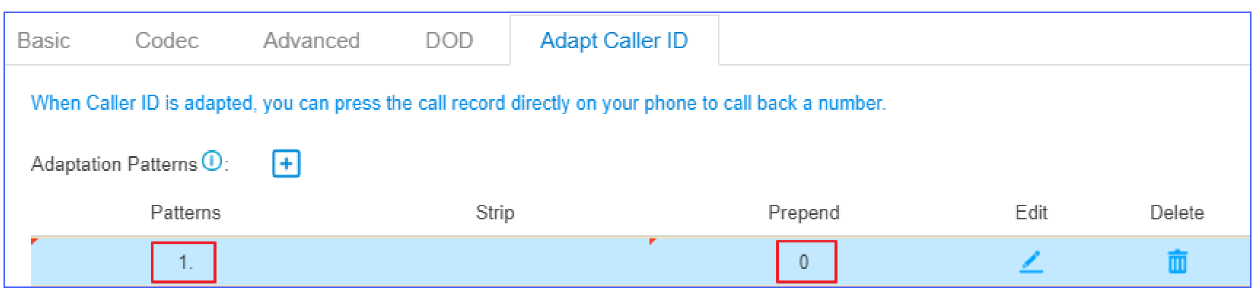
Example 1

Company A wants to add a digit 0 to the 11-digit incoming caller ID number that begins with digit 1 for quick redial purposes.

For example, company A wants to display 012345678910 instead of 12345678910.

In this case, you can configure Adapt Caller ID on trunk 1, and set the rules as follows:

- **Patterns**: 1.
- **Strip**: Leave it blank.
- **Prepend**: 0



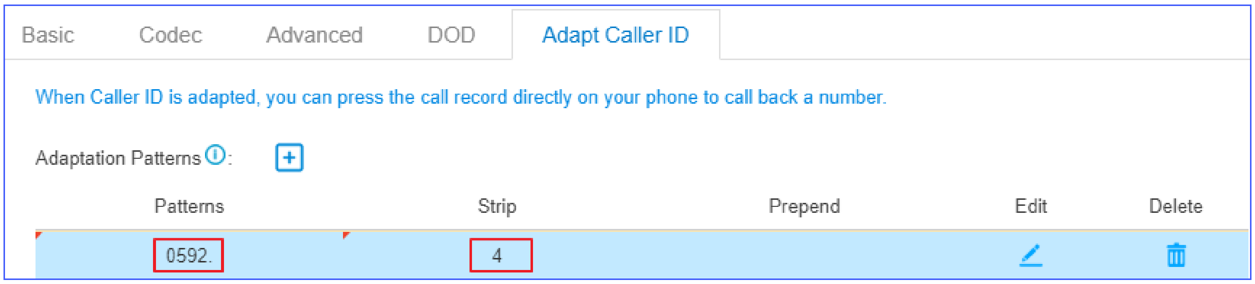
Example 2

Company B wants all Xiamen numbers to be displayed as local number without Xiamen area code (0592) that is received through the trunk 2.

For example, company B wants to display number 5503301 instead of 05925503301.

In this case, you can configure Adapt Caller ID on trunk 2, and set the rules as follows:

- **Patterns**: 0592.
- **Strip**: 4
- **Prepend**: Leave it blank.



Inbound Route Examples

Inbound Route Examples

This topic provides sample configurations that will help you understand DID setting and Caller ID setting of inbound routes.

Note: The following examples ignore time condition, you can set time condition according to your needs.

Inbound route without limit

Any calls to the selected trunk will be routed to the inbound route destination. You can set an inbound route as follows:

- **Name**: Set a name to help you identify it.
- **Member Trunks**: Select desired trunk(s).
- **Destination**: Set the destination.

  Leave all other fields blank.

Inbound route based on a DID number

If a trunk has multiple DID numbers, you can add multiple inbound routes that based on different DID numbers. When users dial different DID numbers, they will be routed to different destinations.

The following example shows an inbound route based on DID number 5503301.

- **Name**: Set a name to help you identify it. For DID routes, you can set the name as the DID number, which helps you identify the route.
- **DID Pattern**: 5503301
- **Member Trunks**: Select the trunk that has the DID number.
- **Destination**: Set the destination.

  Leave all other fields blank.

Inbound route based on consecutive DID numbers

If a trunk has multiple consecutive DID numbers, you can quickly set the DID number range in an inbound route to route calls to different destinations based on the DID numbers.

The following example shows an inbound route based on DID range 5503301-5503305, which will route calls to extension 1001-1005.

- **Name**: Set a name to help you identify it.
- **DID Pattern**: 5503301-5503305
- **Member Trunk**: Select the trunk that has the DID numbers.
- **Destination**: Select **Extension Range**, and enter the extension range 1001-1005.

  Leave all other fields blank.

Inbound route based on Caller ID

By default, PBX routes inbound calls without limit. If you set **Caller ID Pattern**, PBX will route calls only when the users' caller ID numbers match the Caller ID Pattern.

In the following example, the inbound route will route caller ID numbers that start with digit 1 to the destination. For example, number 532352584 that doesn't start with digit 1 can not call in the system through this inbound route.

- **Name**: Set a name to help you identify it.
- **Caller ID Pattern**: *1.*
- **Member Trunks**: Select desired trunk(s).
- **Destination**: Select a destination.

  Leave all other fields blank.

Inbound route based on Caller ID and DID numbers

If you set both **DID pattern** and **Caller ID pattern** for an inbound route, PBX will check if the DID numbers and the user's caller ID number match the DID pattern and Caller ID pattern. Only the matched incoming calls can be routed to the pre-configured destination.

In the following example, when users dial 5503301 with phone number starting with digit 1, the inbound call will be routed to the destination.

- **Name**: Set a name to help you identify it.
- **Caller ID Pattern**: *1.*
- **DID Pattern**: 5503301
- **Member Trunk**: Select desired trunk(s).
- **Destination**: Select a destination.

  Leave all other fields blank.

Route Inbound Calls Based on DID

This topic describes what is DID numbers and how to configure inbound routes on Yeastar Cloud PBX to route inbound calls based on DID.

DID numbers

DID (Direct Inward Dialing) is a telephone service that allows outside users to reach a certain destination instead of going to a receptionist or a queue and needing to dial an extension number.

DID numbers are provided by the trunk provider.

The trunk provider usually assigns a range of numbers to the VoIP trunk or the physical trunk. There is an extra charge for the DID numbers. Contact your trunk provider for more information about DID numbers.

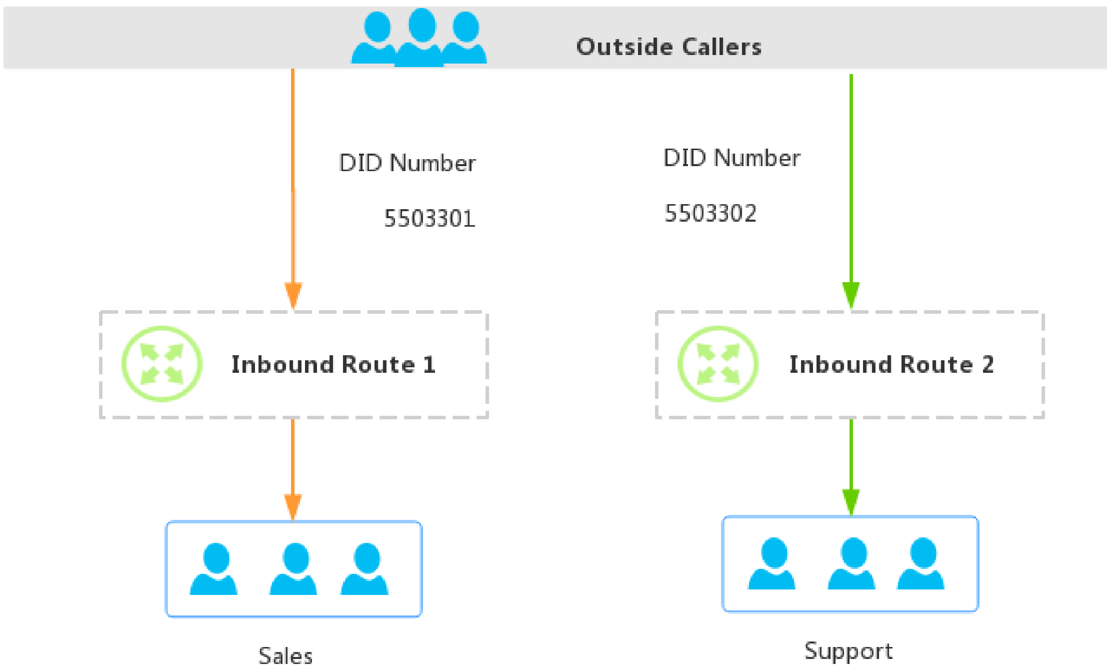
Configure DID routing - single DID

Bind a DID number to an inbound destination.

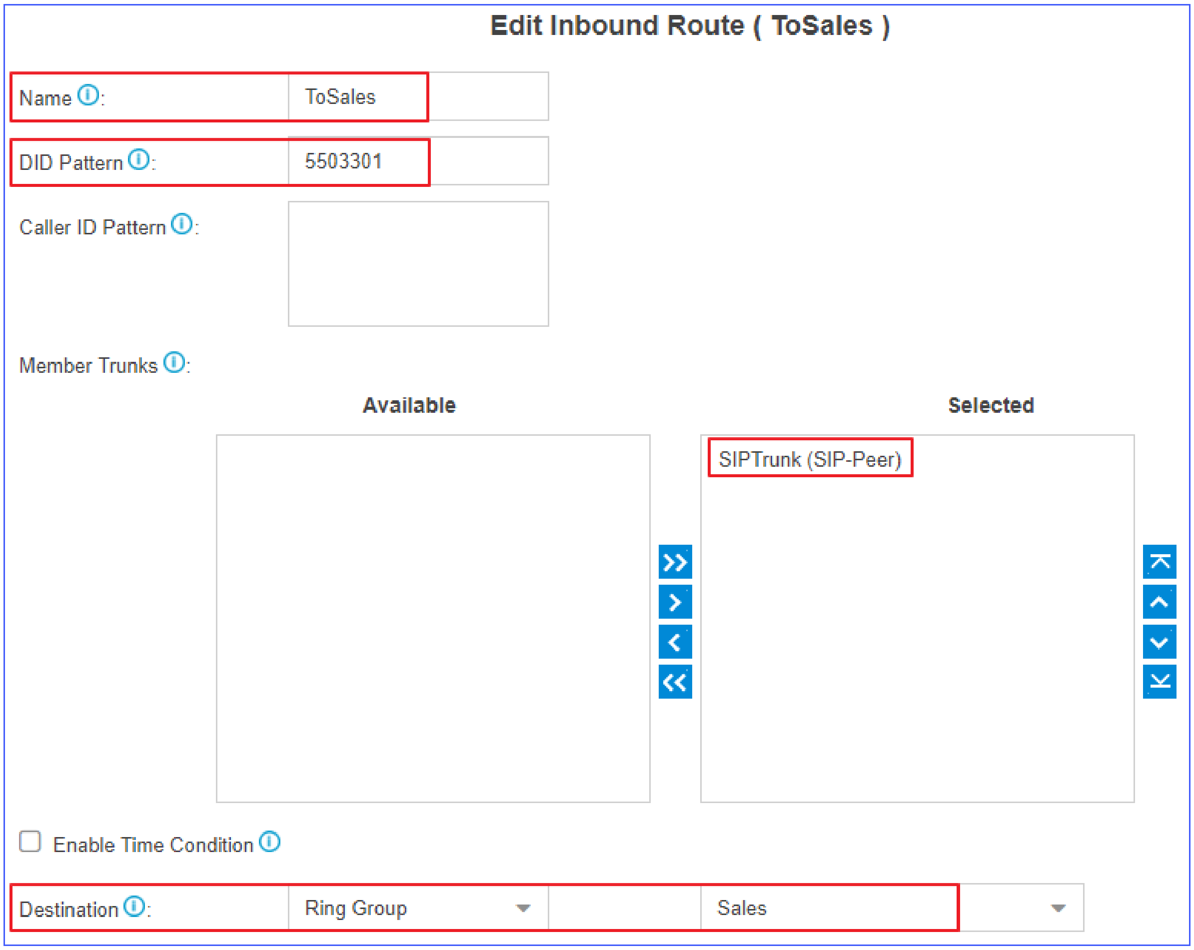
Example:

You purchased two DID numbers from the SIP trunk provider: 5503301 and 5503302.

To route inbound calls to different destinations based on different DID numbers, you can set up two inbound routes for the two DID numbers.

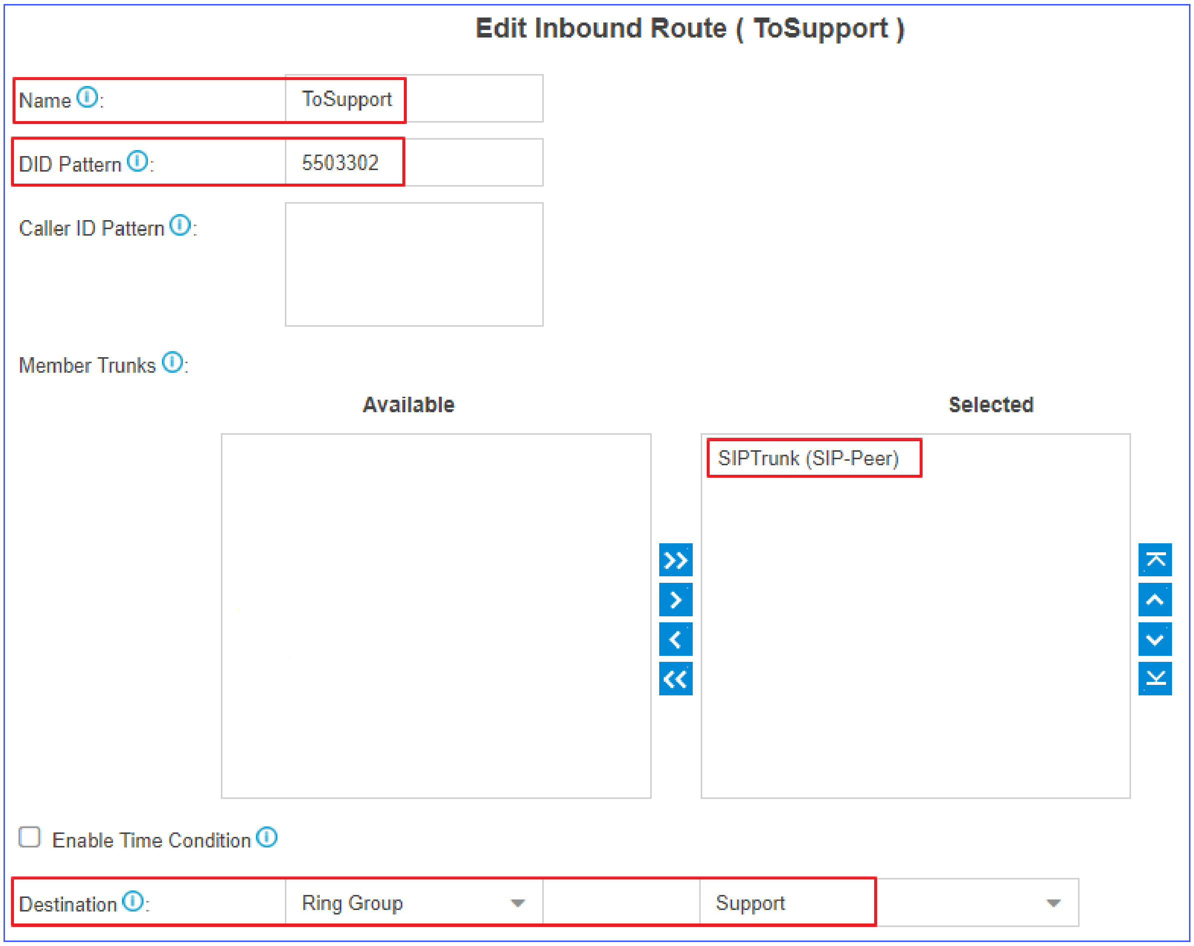


1. Inbound Route **ToSales** for DID number 5503301.



- **Name**: Set a name to help you identify it.
- **DID Pattern**: Enter the DID number *5503301*.
- **Caller ID Pattern**: Leave it blank, which means no limit on caller's Caller ID.
- **Member Trunks**: Select the trunk that is bound with the DID number.
- **Destination**: Select the desired destination. When users dial the DID number 5503301, the call will be routed to the destination.

2. Inbound Route **ToSupport** for DID number 5503302.



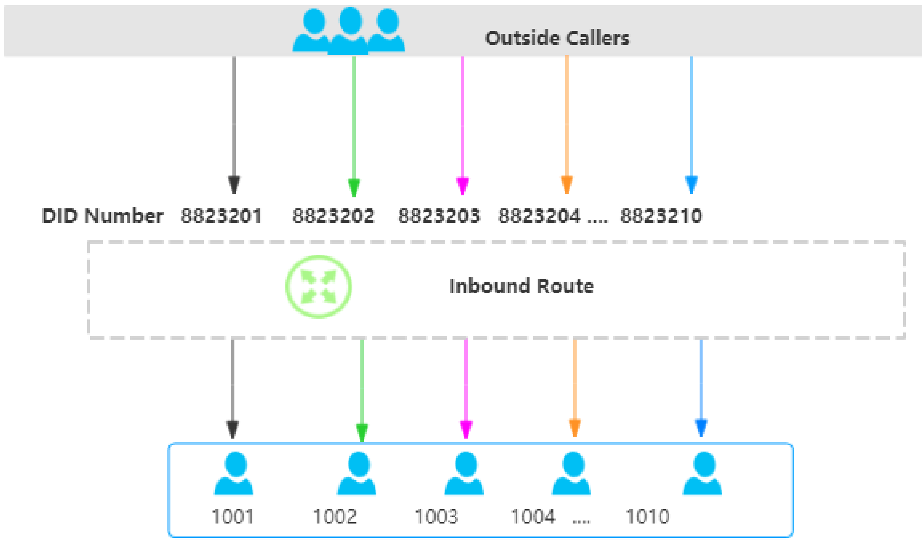
- **Name**: Set a name to help you identify it.
- **DID Pattern**: Enter the DID number *5503302*.
- **Caller ID Pattern**: Leave it blank, which means no limit on caller's Caller ID.
- **Member Trunks**: Select the trunk that is bound with the DID number.
- **Destination**: Select the desired destination. When users dial the DID number 5503302, the call will be routed to the destination.

Configure DID routing - multiple DIDs

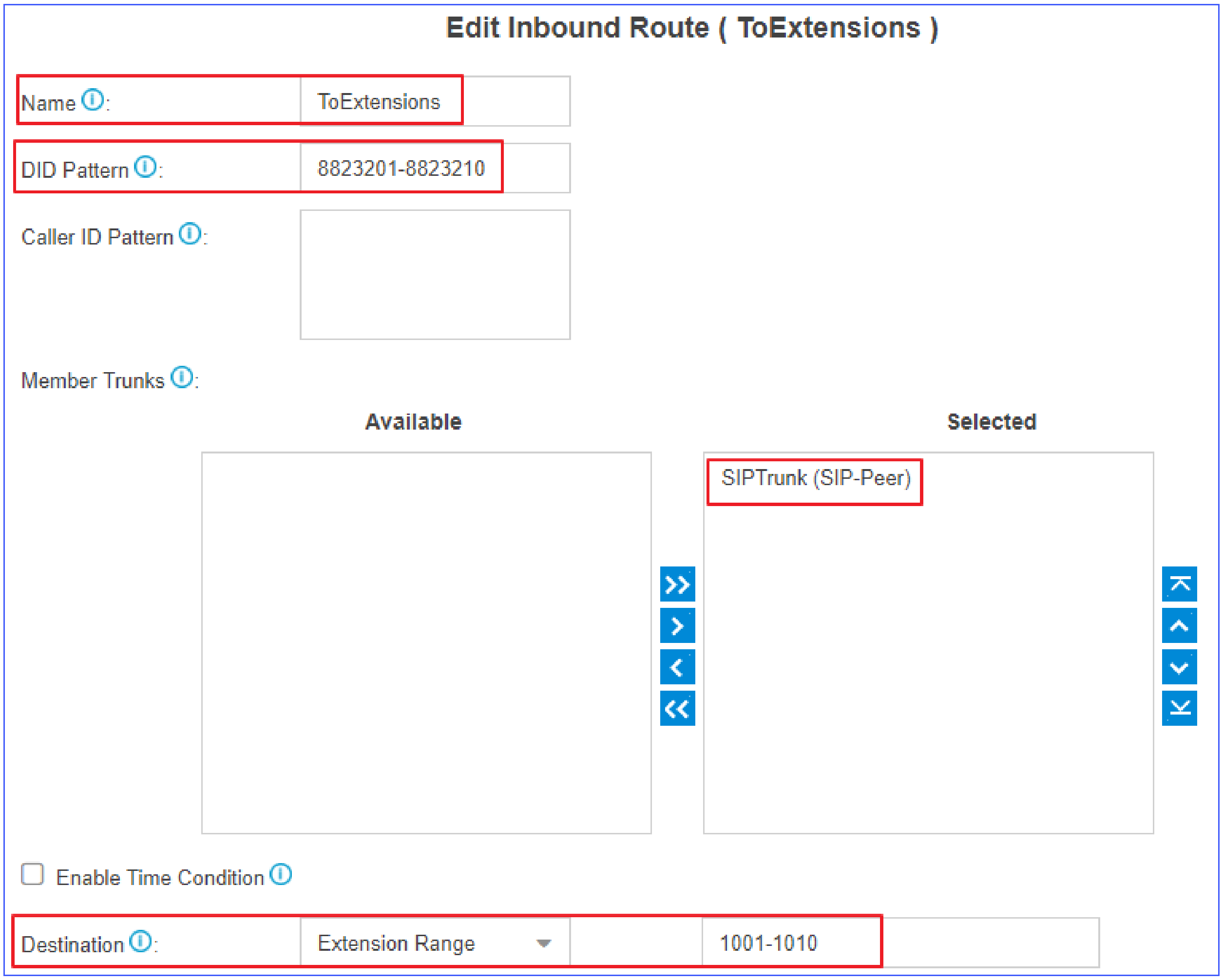
You can assign DID numbers to extension users one by one. When an outside user dials an DID number, the user can reach a specific extension directly.

Note: The DID numbers should be consecutive DID numbers.

Example: You purchased 10 DID numbers from the SIP trunk provider: 8823201-8823210.



To assign the DID numbers one by one to extension 1001-1010 , you can configure the inbound route as follows.



- **Name**: Set a name to help you identify it.
- **DID Pattern**: Enter the DID range *8823201-8823210*.
- **Caller ID Pattern**: Leave it blank, which means no limit on caller's Caller ID.
- **Member Trunks**: Select the trunk that is bound with the DID numbers.
- **Destination**: Select **Extension Range**, and enter the extension range *1001-1010*.

Note: The number of extensions and DID numbers must be the same.

Route Inbound Calls Based on Caller ID

This topic describes what is Caller ID routing and how to configure inbound routes on Yeastar Cloud PBX to route inbound calls based on Caller ID.

Caller ID routing

Caller ID (Caller Identification) is a telephone service that displays a caller's phone number on the called party's phone device before the call is answered.

Caller ID routing allows users to accept or reject calls based on the caller's phone number. Inbound calls which match the Caller ID pattern on PBX will be routed to the pre-configured destination. For those unmatched, calls can not be established.

Scenarios

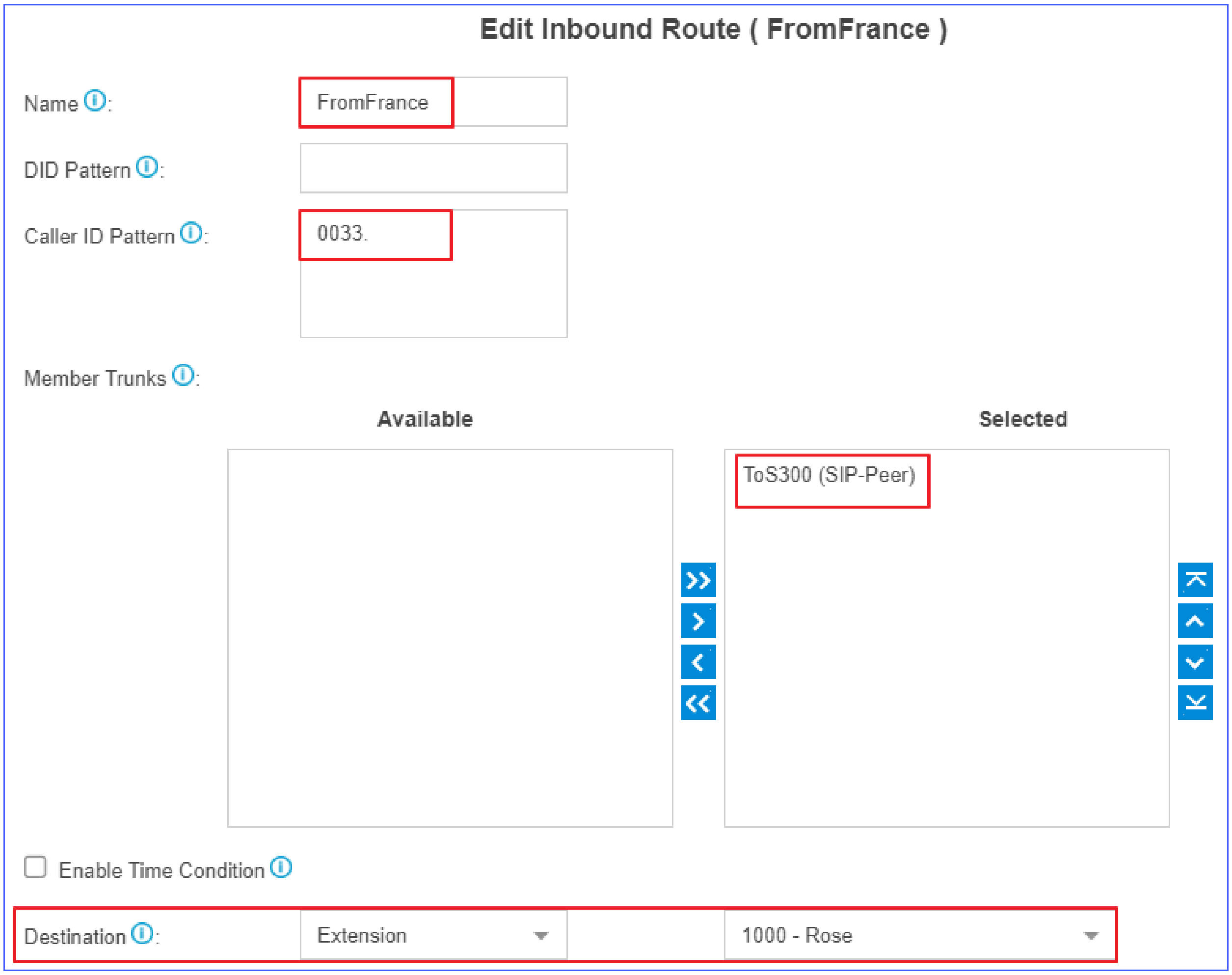
A company is dedicated to offering targeted service for different regions, the company hopes that the Caller ID of inbound calls can be identified and the calls can be routed to responsible employees. In this case, you can set Caller ID patterns for inbound routes.

Configuration Example

Company A assigns pre-sales business in France to Rose, and pre-sales business in America to Mike. Refer to the following table and related configuration figures.

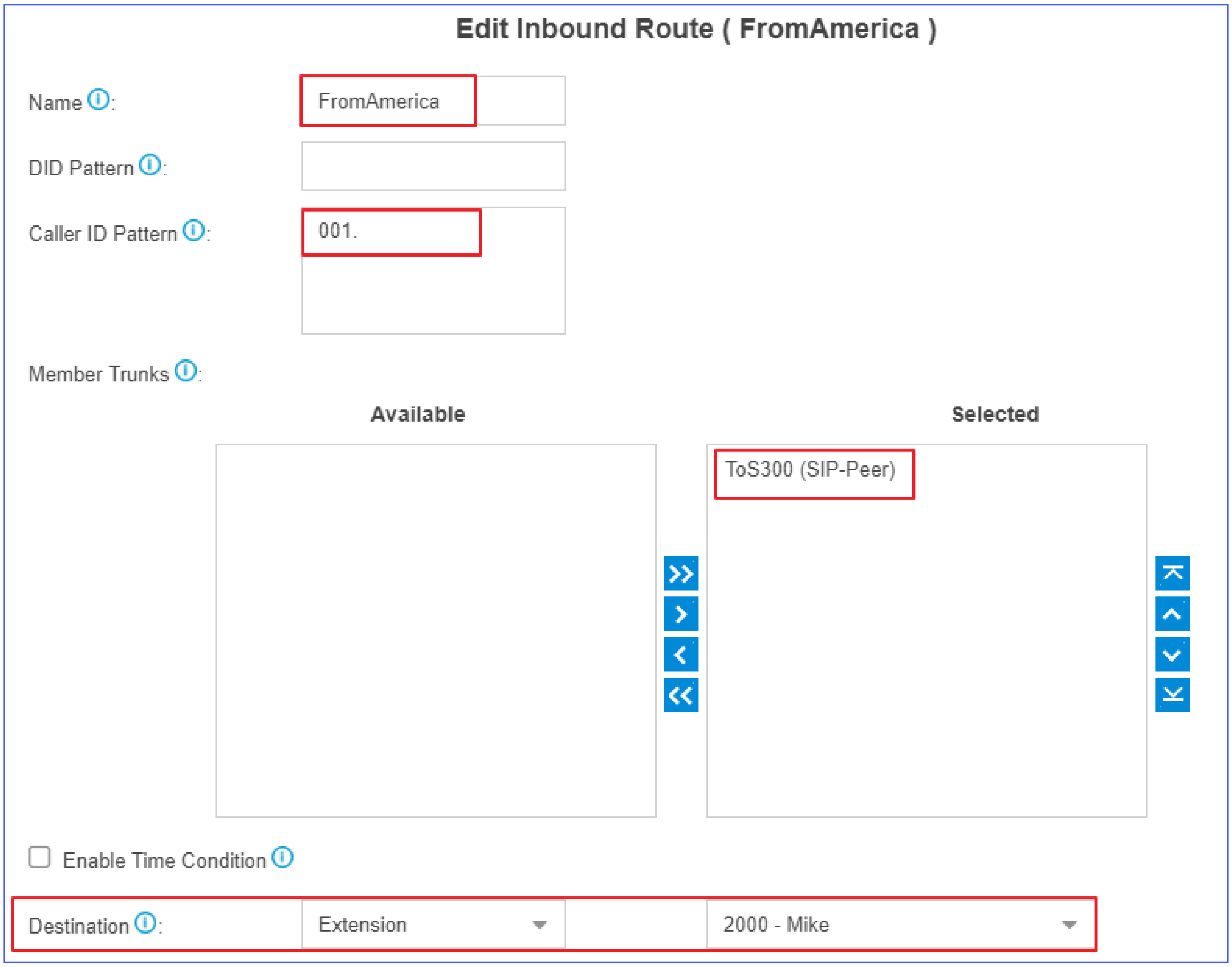
Name	Extension	Responsible Country	Area Code
Rose	1000	France	0033
Mike	2000	America	001

Configure Caller ID pattern for Rose



- **Name**: Set a name to help you identify it.
- **Caller ID Pattern**: Enter the caller ID pattern *0033.*.
- **Member Trunks**: Select the trunk that is bound with the caller ID pattern.
- **Destination**: Select the desired destination. When a caller calls to the trunk with the caller ID starting with 0033, the call will be routed to extension 1000.

Configure Caller ID pattern for Mike



- **Name**: Set a name to help you identify it.
- **Caller ID Pattern**: Enter the caller ID pattern *001.*.
- **Member Trunks**: Select the trunk that is bound with the caller ID pattern.
- **Destination**: Select the desired destination. When a caller calls to the trunk with the caller ID starting with 001, the call will be routed to extension 2000.

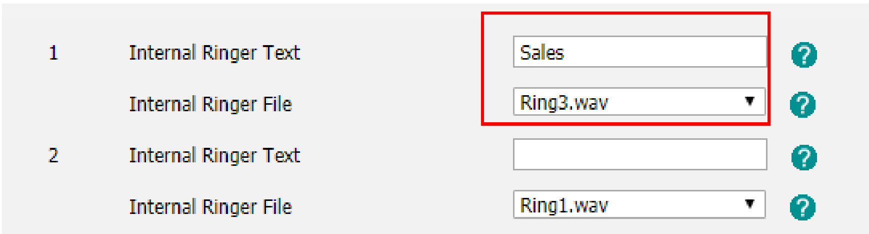
Distinguish Inbound Calls

Distinguish Inbound Calls by Ring Tones

Distinctive ringtone distinguishes calls from different inbound routes. You can set distinctive ringtones on different inbound routes. When a user hears the ringtone of an incoming call, he/she may notice the intention of the call.

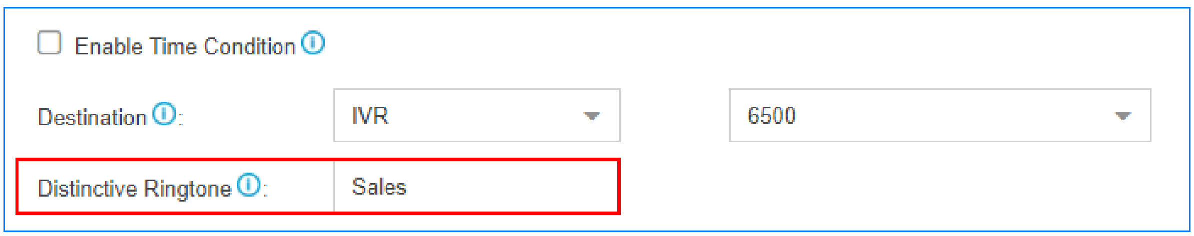
Note: Distinctive Ringtone feature needs support from the IP phones. We take Yealink phone as an example.

1. Log in the phone web interface, go to **Settings > Ring**, select a ringtone and set the name.



   a. In the **Internal Ringer Text** field, enter the ringtone name.
   b. In the **Internal Ringer File** drop-down list, select a ringtone file.
   c. Click **Confirm** to save the settings.

2. Log in the PBX web interface, go to **Settings > PBX > Call Control > Inbound Routes**, select an inbound route to edit.



   a. In the **Distinctive Ringtone** field, enter the ringtone name that is configured on IP phone.
   b. Click **Save** and **Apply**.

When a call comes through the inbound route, the phone will play corresponding ringtone.

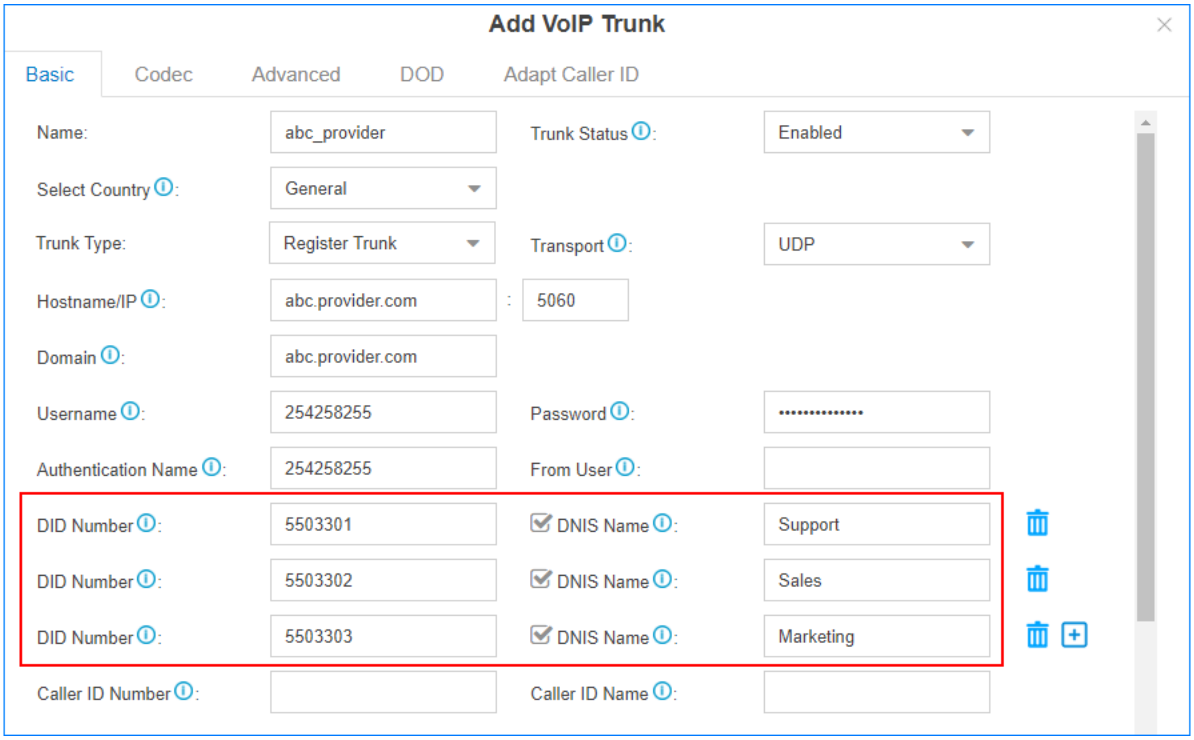
Distinguish Inbound Calls by DNIS Name

DNIS (Dialed Number Identification Service) is used to identify where the incoming call is from. You can set different DNIS names for different trunks or set different DID numbers and DNIS names for a trunk. When external users make outbound calls to PBX, extension users can identify incoming call by DNIS name.

1. Go to **Settings > PBX > Trunks**, click beside the trunk that you want to edit.

2. On the **Basic** page, enter the trunk DID number, and set the **DNIS Name** for the DID number.

   Note: For VoIP Peer Trunk, click **Advanced** tab to find the DID settings.

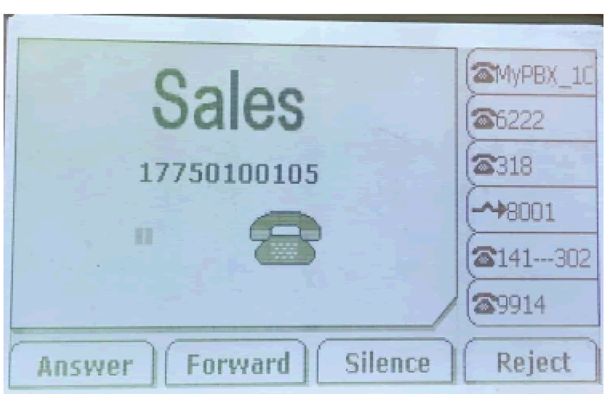


3. If the trunk has another DID number, click [+] to add a DID number and set a **DNIS name**.

   For example, a VoIP trunk has 3 DID numbers. 5503301 for Support, 5503302 for Sales, and 5503303 for Marketing. When external users dial a DID number, extension users can notice the intention by DNIS name displayed on an IP phone.

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

Make a call to the trunk of the PBX, the user who receives the call will see the incoming caller ID and the DNIS name of the trunk.



Distinguish Inbound Calls by Caller ID

When inbound calls are routed from a ring group/queue or an IVR, Yeastar Cloud PBX can display the name of ring group/queue/IVR. When the extension user receives a call from the ring group/queue/IVR, he/she may notice the intention of the inbound call.

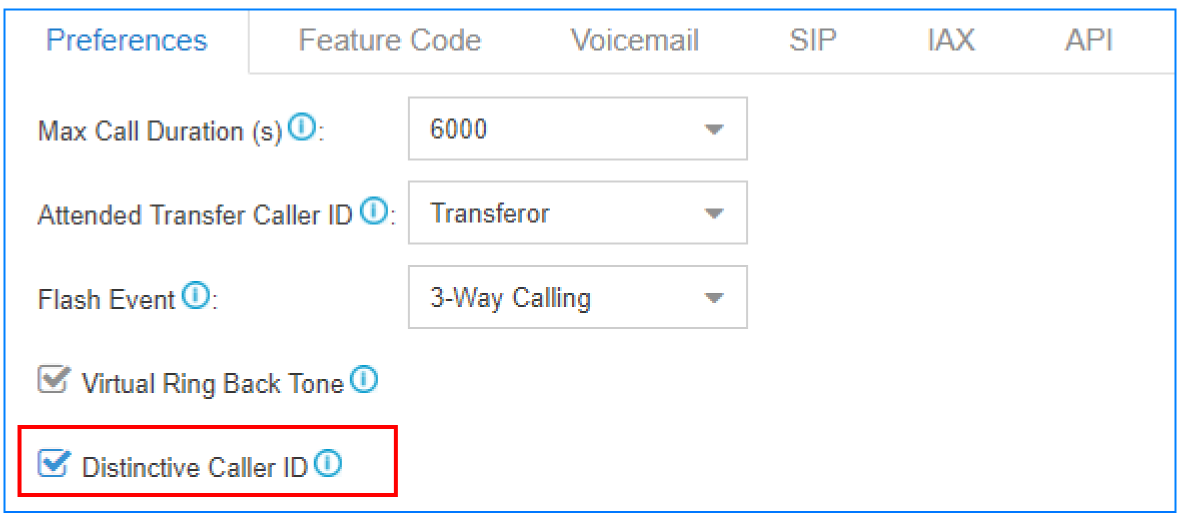
For example:

Set up two Ring Groups according to your organization, one is named as Sales, the other is named as Support.

You can set up two inbound routes to route incoming calls to different destinations by different trunks, and enable **Distinctive Caller ID** feature.

- When external users call to PBX, and IP phones of Sales members ring, "Sales" will be displayed on IP phones.
- When external users call to PBX, and IP phones of Support members ring, "Support" will be displayed on IP phones.

1. Go to **PBX > General > Preferences**, select the checkbox of **Distinctive Caller ID**.



2. Click **Save** and **Apply**.

Outbound Routes

Outbound Route Overview

An outbound route is used to tell the PBX which extension users are allowed to make outbound calls and which trunk to use for the outbound calls.

How does an outbound route work?

Every time user dials a number, PBX will do the following in strict order:

1. Examine the number user dialed.
2. Compare the dialed number with the pattern that you have defined in route 1.
    - If it matches, PBX will route the call out using the associated trunk.
    - If it does not match, PBX will match the number with the pattern that you have defined in route 2, and so on .

Dial Patterns of Outbound Route

This topic describes dial pattern settings of Outbound Route to help you understand and configure the dial patterns of Outbound Route.

Pattern

A pattern specifies routing rules to route a call based on the digits dialed by a user. The PBX matches a dial pattern and routes the call out based on the dial pattern.

Pattern	Description
X	Refers to any digit between 0 and 9.
Z	Refers to any digit between 1 and 9.
N	Refers to any digit between 2 and 9.
[###]	Refers to any digit in the brackets, example [123] would match the numbers 1, 2, or 3. Range of numbers can be specified with a dash, example [136-8] would match the numbers 1, 3, 6, 7, and 8.
.	Wildcard . matches one or more numbers. Example *9011.* matches any numbers starting with 9011 (excluding 9011 itself).
!	• Wildcard *!* matches none or more than one characters. Example *9011!* matches any numbers starting with 9011 (including 9011 itself).

Strip

Strip is an optional setting, it defines how many digits will be stripped from the front of the dialed number before the call is placed.

Example:

If you set **Pattern** as *9.* and set **Strip** as *1*.

If a user wants to call number 1588902923, he/she should dial 91588902923. The PBX will strip digit 9 from the dialed number, and call the number 1588902923.

Prepend

Prepend is an optional setting. The prepend will be added to the beginning of a successful match. If the dialed number matches the **Pattern**, the prepend will be added to the beginning of the number before placing the call.

Example:

If a trunk requires 10-digit dialing, but users are more comfortable with 7-digit dialing, you can prepend a 3-digit area code to all 7-digit phone numbers before the calls are placed.

Related information
Outbound Route Examples

Add an Outbound Route

To allow users to make outbound calls through trunks, you need to set up at least one outbound route on the PBX.

The PBX 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.

1. Go to **Settings > PBX > Call Control > Outbound Routes**, click **Add**.
2. On the configuration page, configure an outbound route according to your needs.
    - **Name**: Enter a name to help you identify it.
    - **Dial Patterns**: Used to match the digits that users dial. When the dialed numbers match a dial pattern, PBX will route the call out through matched outbound route.

Pattern	Description
X	Refers to any digit between 0 and 9.
Z	Refers to any digit between 1 and 9.
N	Refers to any digit between 2 and 9.
[###]	Refers to any digit in the brackets.
.	Wildcard . matches one or more numbers.
!	Wildcard *!* matches none or more than one characters.

- **Member Trunks**: Select a trunk to make outbound calls. If the dialed number matches a dial pattern of the outbound route, PBX will route the call out through selected trunk.
- **Extensions**: Select which extensions are allowed to use this outbound route.
- **Password**: Optional. Set a password for the outbound route. If a password is set, users are required to enter a password when they try to make outbound calls through this route.
    - **None**: No password is needed.
    - **PIN List**: Select a PIN list. Users are required to enter a password in the PIN list when they try to make outbound calls through this outbound route.
    - **Single Pin**: Enter a password. Users are required to enter the password when they try to make outbound calls through this outbound route.
- **Rrmemory Hunt**: Optional.
    - If the feature is enabled, PBX will remember which trunk was used last time, and then use the next available trunk to call out.

      For example, PBX uses the first trunk to call out, then it will use the second trunk to call out next time.
    - If the feature is disabled, PBX will use trunks orderly to call out.
- **Time Condition**: Optional. You can define during which time period can users use this outbound route. By default, users can call out through the outbound route at any time.

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

Note: After you finish the outbound route configurations, you need to check and adjust the priority of your outbound routes, so that PBX can match and route the call out through the proper outbound route.

Related concepts
Dial Patterns of Outbound Route

Related information
Outbound Route Examples

Outbound Route Examples

This topic provides sample configurations that will help you understand dial patterns of outbound route.

Route Name: Domestic

In Xiamen, China, local numbers are all 7-digit numbers and the numbers do not start with 0, such as 5503305.

For long-distance calls, you need to dial the 4-digit area code and local numbers, such as 0595-5503305. The area code in China is in the format of 0ZXX, the first digit is 0, and the second digit cannot be 0.

Pattern	Strip	Prepend	Description
90ZXX.	1	Leave it blank.	This is for a long-distance call. The long-distance number starts with 0, and users should dial 9 before the number. **Note:** Before placing the call, PBX will strip the leading digit 9. **Example**: To call number 05955503303, the user should dial 905955503303.
9ZXXXXXX	1	Leave it blank.	This is for a local call. The local number starts with digit 1-9, and users should dial 9 before the number. **Note:** Before placing the call, PBX will strip the leading digit 9. **Example**: To call number 5503301, the user should dial 95503301.

Route Name: Mobile

All mobile phone numbers in China are 11-digit numbers and start with digit 1, such as 15880260666.

Pattern	Strip	Prepend	Description
1XXXXXXXXXX	Leave it blank.	Leave it blank.	Users can dial the mobile number as they usually do. **Example**: To call number 15880260666, dial 15880260666.

Route Name: International_Call

All international numbers start with digits 00.

Pattern	Strip	Prepend	Description
00.	Leave it blank.	Leave it blank.	Numbers start with digits 00 will go through this outbound route. **Example**: To call number 16262023379, dial 001626202379.

Import Outbound Routes

You can import outbound routes to quickly set up outbound routing on Yeastar Cloud PBX.

1. Go to **Settings > PBX > Call Control > Outbound Routes**, click **Import**.
2. Click **Download the Template**, add the outbound routes information in the template file.

   Note:
   - The imported file should be a UTF-8 `.csv` file.
   - For requirements of the import parameters, refer to Import Parameters - Outbound Routes.

3. Click **Browse** to upload the template file.
4. Click **Import**.

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, greatly saving calling cost for your company.

Note: The route priority is important, especially if there is some overlap. For example, the number 5503305 matches both a dial pattern of `ZXXXXXX` and `X.`, the PBX will send the call through the outbound route with the highest priority.

Example:

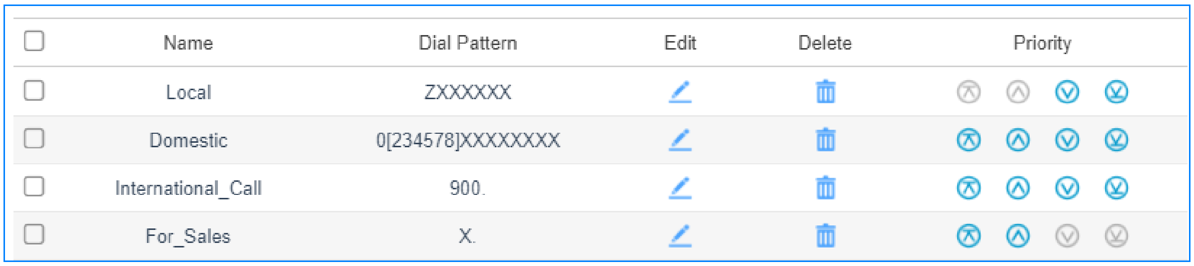
When users dial 05503301, both of the two outbound routes match 05503301:

- 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. Go to **Settings > PBX > Call Control > Outbound Routes**.
2. Click the buttons to adjust the priority of your outbound routes.

Note: PBX will match outbound route from top to bottom.



Name	Dial Pattern	Edit	Delete	Priority
Local	ZXXXXXX			
Domestic	0[234578]XXXXXXXX			
International_Call	900.			
For_Sales	X.			

- : Put this outbound route at the top.
- : Move this outbound route upward.
- : Move this outbound route downward.
- : Put this outbound at the bottom.

Edit an outbound route

1. Go to **Settings > PBX > Call Control > Outbound Routes**.
2. Click beside the outbound route that you want to edit.
3. Edit the outbound route.
4. Click **Save** and **Apply**.

Delete an outbound route

1. Go to **Settings > PBX > Call Control > Outbound Routes**.
2. Click beside the outbound route that you want to delete.
3. On the pop-up window, click **Yes** and **Apply**.

Note: After you delete the outbound route, extension users can not make outbound calls through this outbound route.

Outbound Restriction

Outbound Restriction Overview

Outbound Restriction is used to limit how many outbound calls extension users can make within specified time period.

Scenarios

Avoid toll fraud

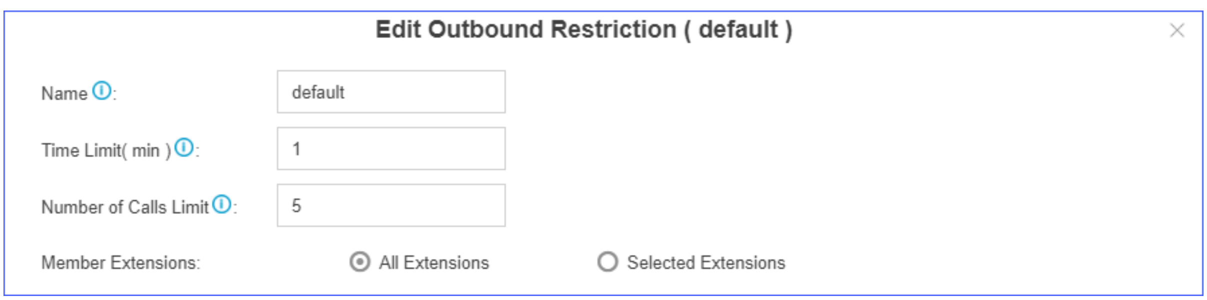
Most toll fraud is committed from the outside. Hackers may attack the system by registering to extensions and making outbound calls frequently.

With the Outbound Restriction rules, if extension users make outbound calls over the limited frequency, the extensions will be blocked and unable to make outbound calls.

Default outbound restriction rule

The PBX has a default rule to limit users to make maximum 5 outbound calls in 1 minute. You can add another Outbound Restriction rule according to your needs.

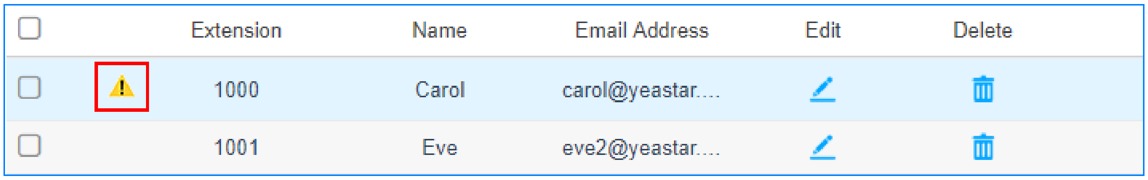
Note: We recommend that you keep the default Outbound Restriction rule.



Cancel restriction of outbound calls

If a user makes outbound calls over the limit, the extension will be locked and prohibited from making outbound calls. On **Extensions** list, the extension status will display .

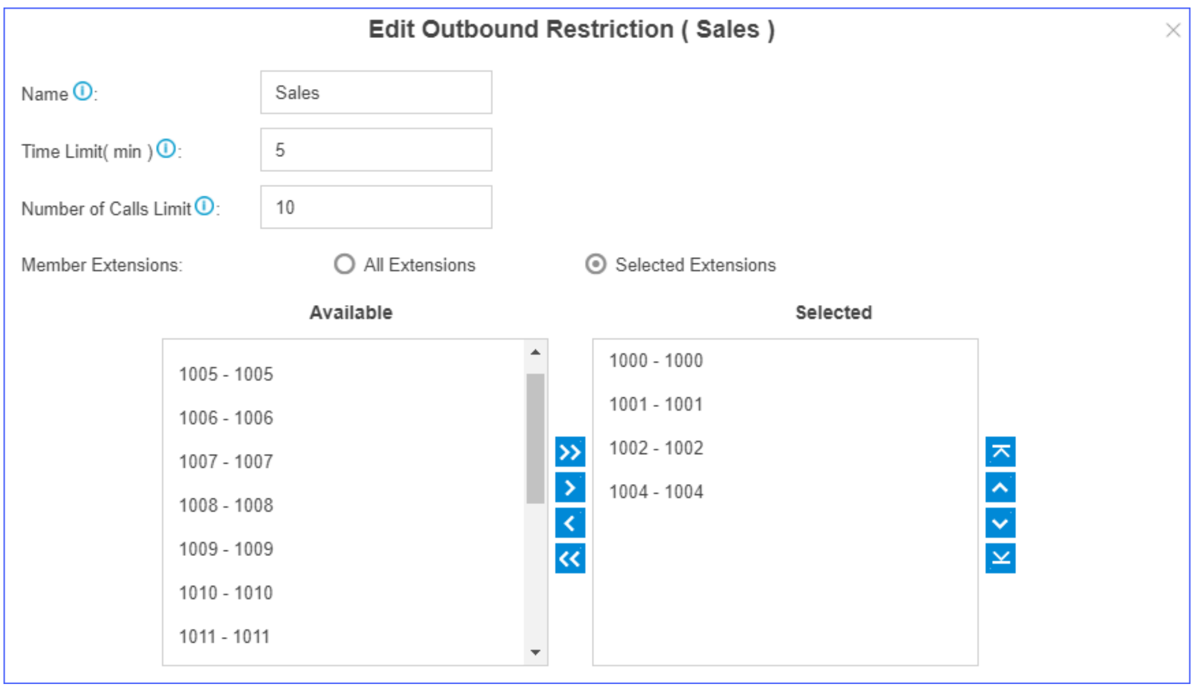
Double click the icon , the extension will be able to make outbound calls again.



Add a Rule to Restrict Outbound Calls

The PBX has a default rule to limit users to make maximum 5 outbound calls in 1 minute. You can add an Outbound Restriction rule to define how many outbound calls the extension users can make during a period of time.

1. Go to **Settings > PBX > Call Control > Outbound Restriction**, click **Add**.
2. On the configuration page, configure an outbound restriction rule according to your needs.



- **Name**: Enter a name to help you identify it.
- **Time Limit(min)**: Set time in minutes to limit the number of outbound calls during the time period.
- **Number of Calls Limit**: Set the number of outbound calls during the specified time period. For example, set **Time Limit(min)** to 5, **Number of Calls Limit** to 10. It means if the selected extension users make outbound calls over 10 times in 5 minutes, the extension(s) will be locked and can not make outbound calls.
- **Member Extensions**: Select extensions which will be restricted by the rule.

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

Manage Outbound Restriction Rules

After you create restriction rules, you can edit or delete them.

Edit an outbound restriction rule

1. Go to **Settings > PBX > Call Control > Outbound Restriction**.
2. Click beside the outbound restriction rule that you want to edit.
3. Edit the outbound restriction rule.
4. Click **Save** and **Apply**.

Delete an outbound restriction rule

1. Go to **Settings > PBX > Call Control > Outbound Restriction**.
2. Click beside the outbound restriction rule that you want to delete.
3. On the pop-up window, click **Yes** and **Apply**.

AutoCLIP Routes

AutoCLIP Overview

AutoCLIP (Auto Calling Line Identity Presentation) is an intelligent call matching feature. You can configure AutoCLIP to route inbound calls to original extensions, which will promote your customer satisfaction and work efficiency.

Scenarios

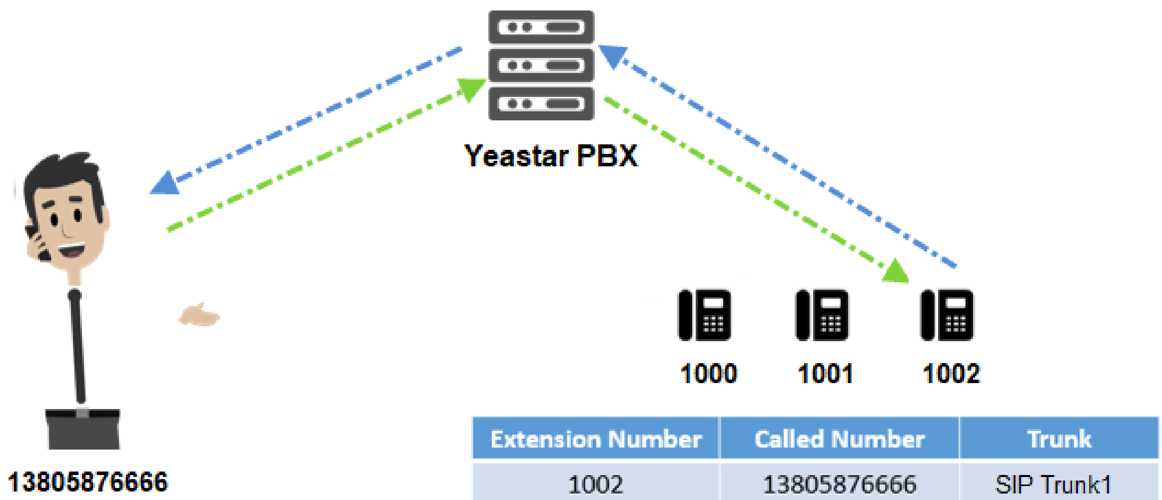
Assume sales representatives in your company often make outbound calls to customers for promotion. More or less, some customers may miss the calls. When customers call back, the calls are routed to the reception or business auto attendant. Neither reception/business auto attendant nor the customers know 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.

How does the PBX redirect calls to original extensions?

1. When extension users make outbound calls, the PBX automatically stores the records to AutoCLIP routing table.
2. When customers call in the PBX, PBX will search the phone numbers from the AutoCLIP routing table.

- If there're matched records in AutoCLIP routing table, the calls will be routed to corresponding extensions.
- If there're not matched records in AutoCLIP routing table, the calls will be routed to the destination specified in inbound routes.



Configure AutoCLIP to Route Inbound Calls to Original Extensions

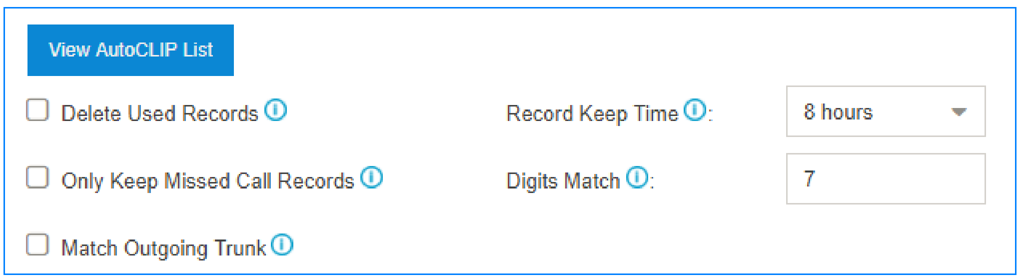
With AutoCLIP feature on Yeastar Cloud PBX, the PBX can route inbound calls from customers to original extensions users who placed the calls. This intelligent call matching feature can greatly improve work efficiency and customer satisfaction.

Note:

- Enable caller ID feature for the trunk that you want to configure AutoCLIP routes, or the PBX can not distinguish the caller ID and perform AutoCLIP.
- If many extension users 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.

1. Go to **Settings > PBX > Call Control > AutoCLIP Routes**.
2. In the **Member Trunks** field, select the trunk(s) from **Available** box to the **Selected** box.

3. Configure the AutoCLIP settings according to your needs.



- **Delete Used Records**: Select this option, PBX will perform AutoCLIP as follows:
    a. When receiving an external call from customer A, the PBX will search the record from AutoCLIP list, and redirect the call to the original extension user that placed the call.
    b. PBX will delete the AutoCLIP record.
    c. When receiving an external call from customer A again, PBX will always route the call to the destination specified by the inbound route instead of searching the record from AutoCLIP list.
    d. If extension users of PBX make outbound calls to customer A again, PBX will generate AutoCLIP record again.

    Note: To restrict PBX from routing all inbound calls from a certain customer to the same extension user, select **Delete Used Records**.

- **Record Keep Time**: Set how long records can be kept in AutoCLIP list. If keep time of a certain record over the value, PBX will automatically delete the record.
- **Only Keep Missed Call Records**: Select this option. Only unconnected outbound calls (missed calls on the called party) will be recorded in AutoCLIP list.

- **Digit Match**: The default value is 7, which means if the digit of caller ID is less than or equal to 7, the PBX will match the whole phone number with all phone numbers in AutoCLIP list. If the digit of caller ID over 7, the PBX will match the last 7 digits of phone number with all phone numbers in AutoCLIP list.

Example:

a. Extension user 2000 makes an outbound call to customer 15880270666, and an AutoCLIP record is generated.
b. When the customer calls in the PBX, the caller ID displays +8615880270666, where +86 stands for country code. To make sure the PBX can exactly match the phone number in AutoCLIP list, you should set **Digit Match** to *11*.
c. If the last 11 digits of +8615880270666 exactly match the phone number in AutoCLIP list, the PBX will route the call to extension 2000.

- **Match Outgoing Trunk**: Select this option. The PBX will route the call to the original extension only when the trunk number dialed by external users matches the trunk that used to place the call earlier.

  Example:

  Extension user (1000) uses trunk1 to call external user (15880273600). PBX will route the call to extension (1000) only when the external user (15880273600) calls the phone number of trunk1.

4. Click **Save** and **Apply**.
5. Test AutoCLIP routes.

   Extension user uses the trunk with AutoCLIP feature to call external users out.

PBX generates an AutoCLIP record when extension user uses the trunk with AutoCLIP feature to call external users out. On the **AutoCLIP Routes** page, click **View AutoCLIP List** to view AutoCLIP record.

SLA Stations

SLA Overview

SLA (Shared Line Appearance) feature helps users share and monitor SIP trunks. After enabling SLA feature for a trunk, the trunk works as the exclusive line for SLA station and is unavailable in both inbound routes and outbound routes.

SLA trunk refers to the trunk with SLA feature enabled. SLA station refers to an extension which is bound with a SLA trunk.

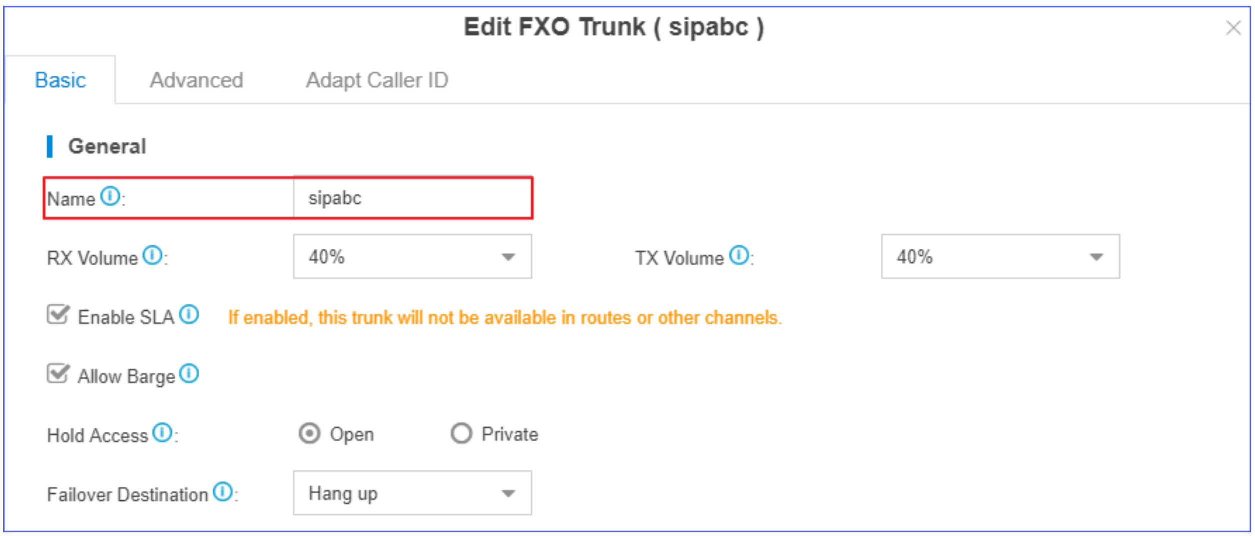
- When an SLA station makes an outbound call through SLA trunk, other members sharing the SLA trunk can monitor the trunk state by BLF keys LED on phone devices.
- When receiving an external call from SLA trunk, all extensions sharing the SLA trunk will ring.

Note: If **Allow Barge** feature is enabled on an SLA trunk, all members can place and join multi-party calls.

SLA Sample Configuration

In a boss-assistant scenario, sometimes assistant needs to answer calls for the boss. So boss and assistant need to share a trunk. In this topic, we introduce how to configure SLA trunk and SLA station on Yeastar Cloud PBX based on a boss-assistant scenario.

Assume that the boss’s phone is extension 2000 and the assistant's phone is extension 1000. The shared trunk name is "sipabc" and the trunk number is 5503305.

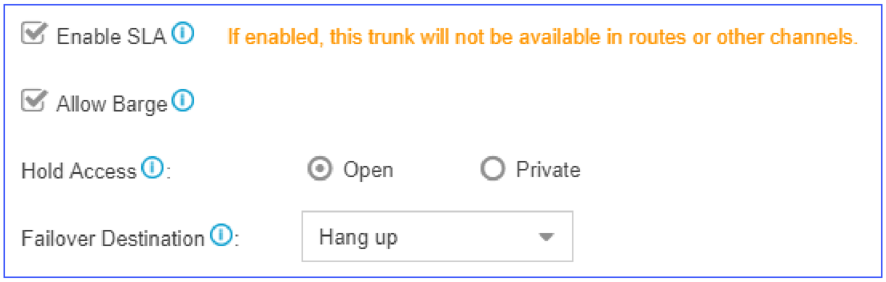


Note: SLA feature should be used in conjunction with BLF keys on phone devices.

You can set up a shared trunk as follows.

1. Enable SLA feature.

    a. Go to **Settings > PBX > Trunks**, click beside the trunk that you want to enable SLA.
    b. On the **Basic** page, select **Enable SLA** and configure the SLA settings.



- **Enable SLA**: Select this option to enable SLA on the trunk.
- **Allow Barge**: Optional. Whether to allow other SLA stations that share the trunk to join the ongoing call by pressing the BLF key on phone devices.
- **Hold Access**: Whether to allow any SLA stations to retrieve a call that's put on hold.
    - **Open**: Any SLA stations that share the trunk can retrieve the call.
    - **Private**: The call can be retrieved only by the SLA station that previously put the call on hold.
- **Failover Destination**: The unanswered calls will be routed to the destination.
    - Hang up
    - Extension
    - Voicemail
    - IVR
    - Ring Group
    - Queue

c. Click **Save** and **Apply**.

2. Add two SLA stations for the same SLA trunk. One SLA station for the boss's extension 2000, the other SLA station for the assistant's extension 1000.

    a. Go to **Settings > PBX > Call Control > SLA**, click **Add**.

    b. On the SLA Station configuration page, set SLA station for the boss.

- **Station Name**: Set a name to help you identify it.
- **Station**: Select the boss's extension 2000.
- **Associated SLA Trunks**: Select SLA trunk from the **Available** box to the **Selected** box.
- **Ring Timeout(s)**: Set the timeout in seconds. When receiving an inbound call, the phone of the SLA station will ring until timeout. The default value is 30s.
- **Ring Delay(s)**: Set the time delay in seconds. Phone of the SLA station will delay ringing after the time defined. The time of **Ring Delay(s)** can not be longer than the time of **Ring Timeout(s)**. The default value is 0s.
- **Hold Access**:Whether to allow any SLA stations to retrieve a call that's put on hold.
  - **Open**: Any SLA stations that share the line can retrieve the call.
  - **Private**: The call can be retrieved only by the SLA station that previously put the call on hold.

c. Click **Save** and **Apply**.

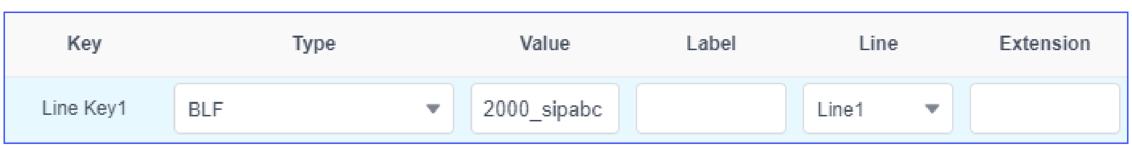
d. Repeat steps **a** to **c** to set the other SLA station for the assistant.

Note: In the **Station** field, select the assistant's extension 1000.

3. On the boss's IP phone (extension 2000), configure a BLF key to monitor SLA trunk.

Note: We take an Yealink IP phone as an example.

a. Log in the phone web interface, go to **DSS key > Line Key** to set a BLF key for the boss.
b. Select a key to configure.



Key	Type	Value	Label	Line	Extension
Line Key1	BLF	2000_sipabc		Line1	

- **Type**: Select **BLF**.
- **Value**: Enter `{ext_num}_{trunk_name}`. In this example, enter *2000_sipabc*.

  Note:
  - `{ext_num}` stands for extension number.
  - `{trunk_name}` stands for trunk name.

- **Line**: Select the line which the extension registers to.
- **Extension**: Optional. You can enter the key name to help you identify it.

c. Click **Confirm**.

4. On the assistant's IP phone (extension 1000), configure a BLF key to monitor SLA trunk.

Note: We take an Yealink IP phone as an example.

a. Log in the phone web interface, go to **DSS key > Line Key** to set a BLF key for the assistant.
b. Select a key to configure.

Key	Type	Value	Label	Line	Extension
Line Key1	BLF	1000_sipabc		Line1	

- **Type**: Select **BLF**.
- **Value**: Enter `{ext_num}_{trunk_name}`. In this example, enter *1000_sipabc*.

Note:
  - `{ext_num}` stands for extension number.
  - `{trunk_name}` stands for trunk name.

- **Line**: Select the line which the extension registers to.
- **Extension**: Optional. You can enter the key name to help you identify it.

c. Click **Confirm**.

If the configuration is correct, you can see the BLF key LED is on.

- **Green**: The trunk is available.
- **Red**: The trunk is busy.

The boss and assistant can share the trunk by SLA.

Related concepts
[Share Trunks by SLA]

Share Trunks by SLA

After setting up SLA stations on PBX and configuring BLF keys on IP phones, users can monitor SLA trunks, receive calls from SLA trunks, and make outbound calls through SLA trunks.

Make outbound calls

SLA station can monitor the status of SLA trunk according to BLF keys status.

Note: For different phone models, there may be some difference in the status of BLF keys.

- If the BLF key used to monitor SLA trunk turns green, it indicates that the trunk is available, and the associated SLA station can make outbound calls through this trunk.

  To make outbound calls, the SLA station should press BLF key first, and dial the external number out after hearing a dial tone.
- If the BLF key used to monitor SLA trunk turns red, it indicates that the trunk is in use. Other SLA stations can not use the trunk to make outbound calls now.

Handle incoming calls

When an external call reaches the SLA trunk, all phones of associated SLA stations will ring, and BLF keys on phone devices will flash in red. Any SLA stations can answer the call by pressing BLF keys.

Barge-in an active call

If **Allow Barge** is enabled for an SLA trunk, other SLA stations are allowed to join an active call.

When an SLA station is in a call with other users using this SLA trunk, other SLA stations can join the active call by pressing the BLF key.

Hold and retrieve calls

During the call, the SLA station can press the BLF key to hold and retrieve the call. Whether an SLA station can retrieve a call or not depends on the **Hold Access**.

Note: **Hold Access** of SLA station has a higher priority than the **Hold Access** of a trunk.

- If **Hold Access** is set to **Open**, other stations that share the trunk can press BLF key to retrieve the call.
- If **Hold Access** is set to **Private**, the call can be retrieved only by the station that previously put the call on hold.

Related concepts
SLA Sample Configuration

Call Features

IVR

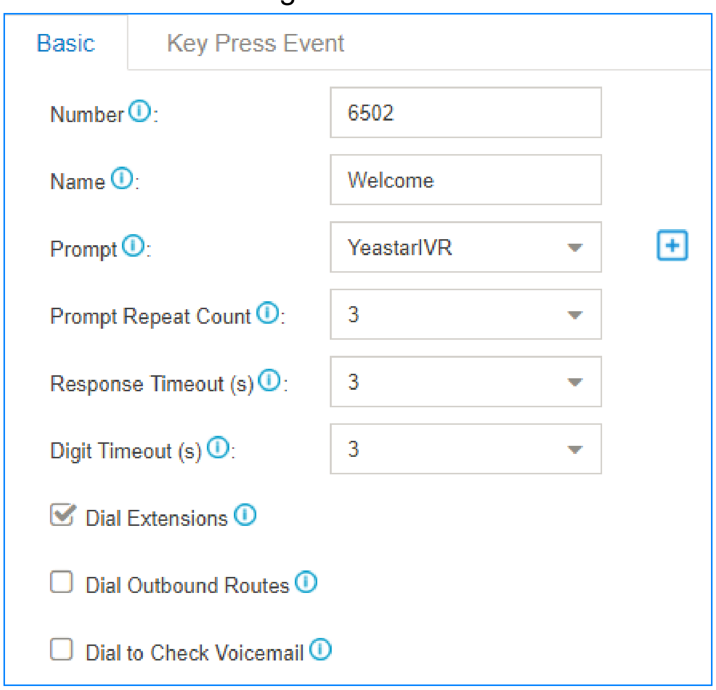
Like most organisations, where possible, we would like to route incoming calls an Auto Attendant. You can create one or more IVR (Auto Attendant) on the system to achieve it.

When calls are routed to an IVR, the system will play a recording prompting them what options the callers can enter such as “Welcome to XX, for sales press 1, for Technical Support press 2”.

Set up an IVR

Set up your own IVR if you need to routing incoming calls via an auto attendant.

1. Go to **Settings > PBX > Call Features > IVR**, click **Add** to add an IVR or edit the default IVR.
2. Edit the **Basic** settings of the IVR.



- **Number**: PBX treats IVR as an extension; you can dial this extension number to reach the IVR from internal extensions.
- **Name**: Set a name for the IVR.
- **Prompt**: Use the default IVR prompt or select your custom IVR prompt.
- **Prompt Repeat Count**: Set how many times the prompt will be played.
- **Response Timeout**: Set how long the PBX will wait for the caller to operate.
- **Digit Timeout**: After the user enters a digit, the user needs to enter the next digit within the timeout.

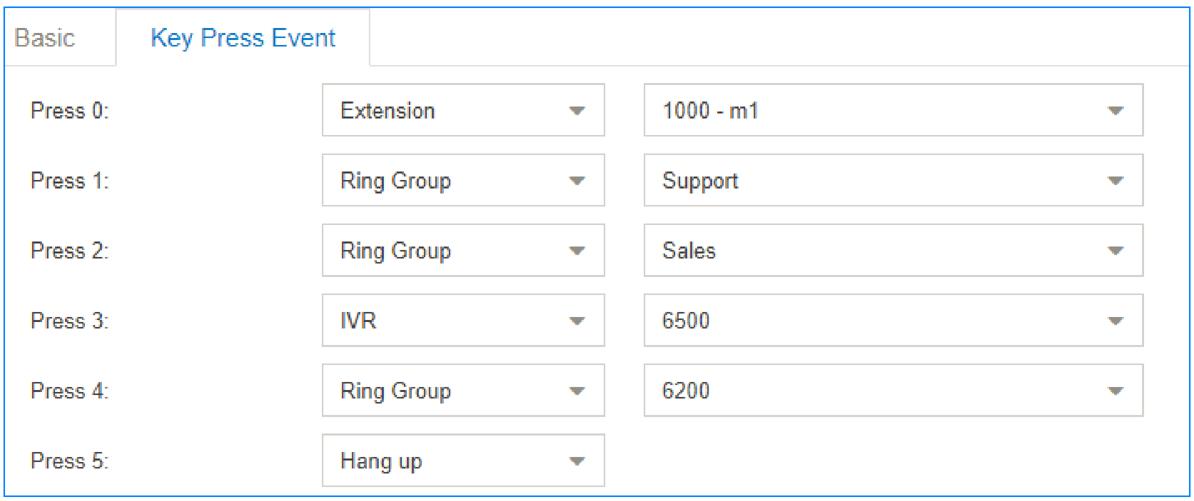
- **Dial Extensions**: Whether to allow callers to dial extension numbers via IVR.
- **Dial Outbound Routes**: Whether to allow callers to dial outbound calls via IVR.

  Note: This option is useful if you interconnect two PBXs. The callers can dial the other PBX's extension number via the IVR. In this solution, you need to configure the appropriate outbound route and inbound route in both of the two connected PBXs.

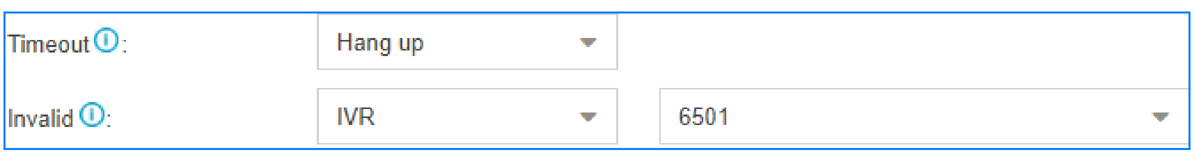
- **Dial to Check Voicemail**: Whether to allow users to check voicemail via IVR.

  Note: This option is for the users who work out of the office. They can call in the PBX and check their voicemail messages via the IVR.

3. Click **Key Press Event** tab, set the destination based on callers key presses.



4. On the **Key Press Event** page, set the **Timeout** destination and the **Invalid Destination**.



   - **Timeout**: If callers do not make an entry within the **Prompt Repeat Count**, they will be transferred to the **Timeout** destination.
   - **Invalid**: If callers enter a digit that is not defined in the IVR, they will be transferred to the **Invalid** destination.

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

Set an IVR Prompt

When users call in the PBX IVR, the users would operate following by the IVR prompt. The PBX system has one default IVR prompt, you can change the IVR prompt to your audio file.

1. Upload a custom prompt or record a custom prompt on the PBX web interface.
2. Go to **Settings > PBX > Call Features > IVR**, edit your IVR.
3. Select the **Prompt** to your custom prompt.
4. Set the **Prompt Repeat Count**.
5. Click **Save** and **Apply**.

Related tasks
Upload a Custom Prompt
Record a Custom Prompt
Convert Audio Files Online
Convert Audio Files via WavePad

Change IVR Prompt Clip

If you need to change one audio clip in the IVR prompt frequently. You can devide your IVR prompt to mutiple audio clips, and change the desired audio clip when you need to change the IVR prompt.

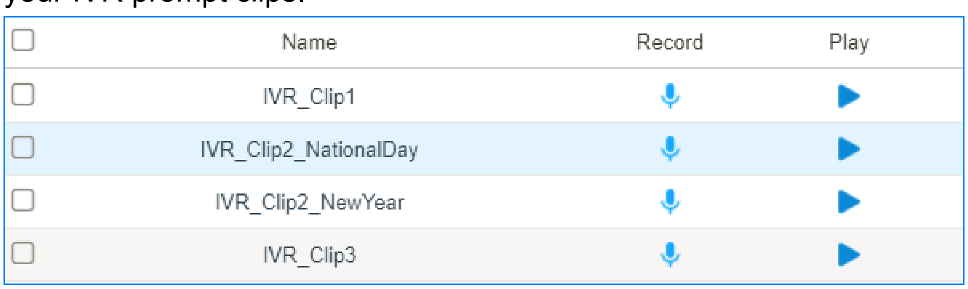
For example, your IVR prompt is like the following:

" Thank you for calling Yeastar. We are currently closed in observance of `Holiday Name`. We will return on `Date`. If you got something urgent, please press 1 to contact our support. To leave a voicemail, please press 2."

The second sentence is what your would change frequently. You can divide the IVR prompt to 3 clips.

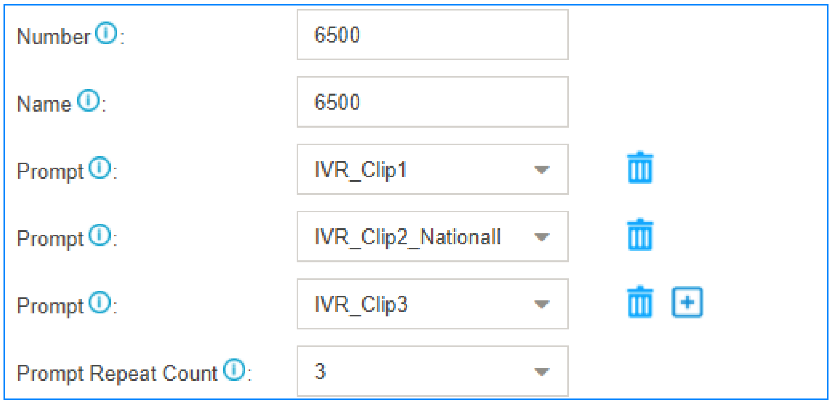
- Clip 1: Thank you for calling Yeastar.
- Clip 2: We are currently closed in observance of `Holiday Name`. We will return on `Date`
- Clip 3: If you got something urgent, please press 1 to contact our support. To leave a voicemail, please press 2.

1. Go to **Settings > PBX > Voice Prompts > Custom Prompts**, click **Upload** to upload your IVR prompt clips.



2. Go to **Settings > PBX > Call Features > IVR**, edit your IVR.

3. Select the **Prompt** to the IVR prompt clip1.
4. Click , and select the **Prompt** to your IVR prompt clip2.
5. Click , and select the **Prompt** to your IVR prompt clip3.



6. Click **Save** and **Apply**.

Next time, when you want to change the IVR prompt, you can change the desired prompt clip instead of changing the whole IVR prompt.

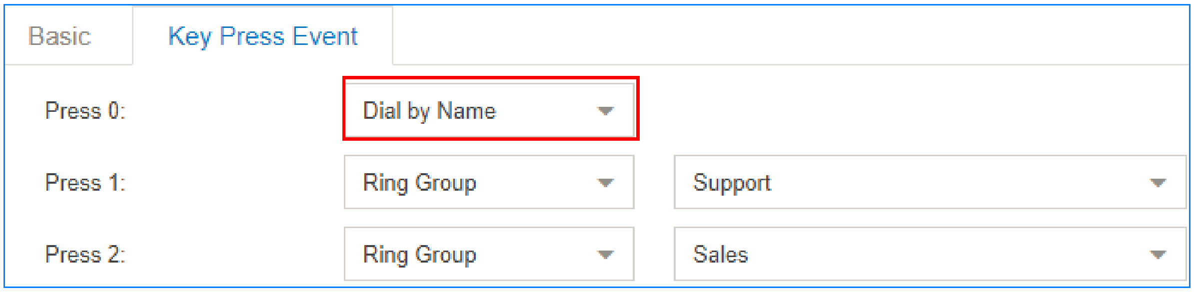
Dial by Name

You can set the IVR Keypress to "Dial by Name", that will allow the callers to find the person by entering the first 3 letters of extensions' first name.

To use **Dial by Name**, you need to do the followings:

- Specify names for extensions on the PBX.
- Better to instruct the callers to use the feature in the IVR prompt.

1. Go to **Settings > PBX > Call Features > IVR**, edit your IVR.
2. Click **Key Press Event** tab, set a key action to **Dial by Name**.



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

Ring Group

A ring group helps you to ring a group of extensions in a variety of ring strategies. For example, you could define all the technical support guys' extensions in a ring group and ring the support guys one by one.

Add a Ring Group

1. Go to **Settings > PBX > Call Features > Ring Group**, click **Add**.

2. Configure the ring group.
    - **Number**: Use the default number or change the number.
    - **Name**: Give a name for the ring group to help you identify it.
    - **Ring Strategy**:
        - **Ring All Simultaneously**: Ring all the available extensions simultaneously.
        - **Ring Sequentially**: Ring each extension in the group one at a time.
    - **Seconds to ring each member**: Define how long the system will wait to ring next member.
    - **Members**: Select the desired extensions to the **Selected** box.
    - **Failover Destination**: Define what will happen if none of the members in the ring group answer the call in the defined time.

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

Queue

Queues are designed to receiving calls in a call center.

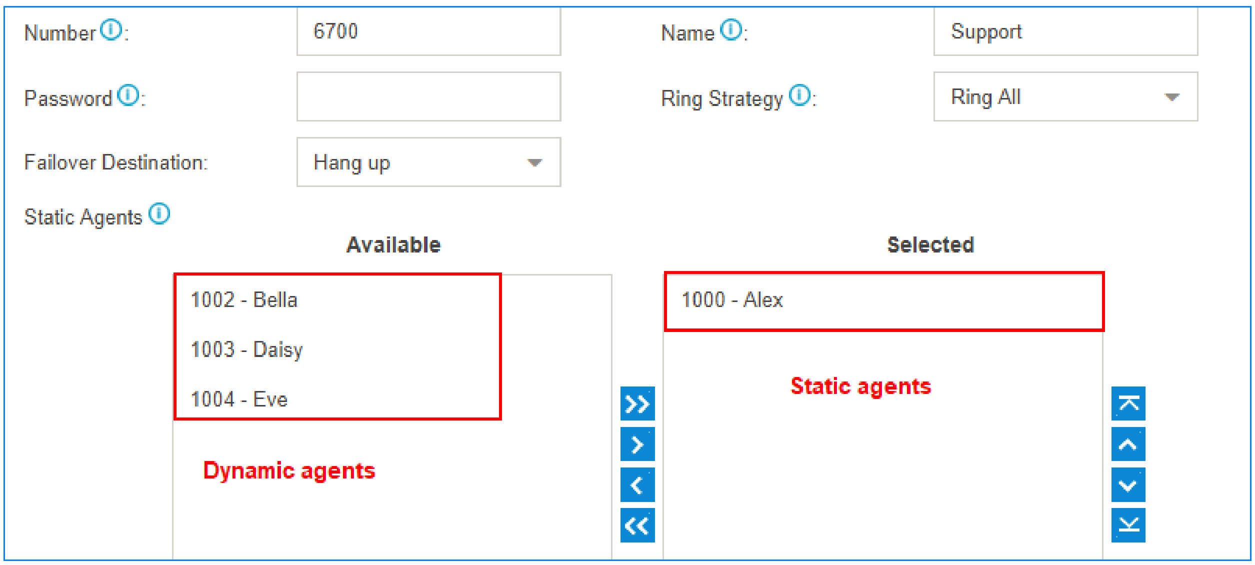
A queue is like a virtual waiting room, in which callers wait in line to talk with the available agent. Once the caller called in PBX and reached the queue, he/she will hear hold music and prompts, while the queue sends out the call to the logged-in and available agents. A number of configuration options on the queue help you to control how the incoming calls are routed to the agents and what callers hear and do while waiting in the line.

Queue Agents

Yeastar Cloud PBX supports dynamic agents and static agents.

- **Static Agent**: A static agent always stays in a queue to receive incoming calls.
- **Dynamic Agent**: A dynamic agent can log in a queue or log out a queue at any time.

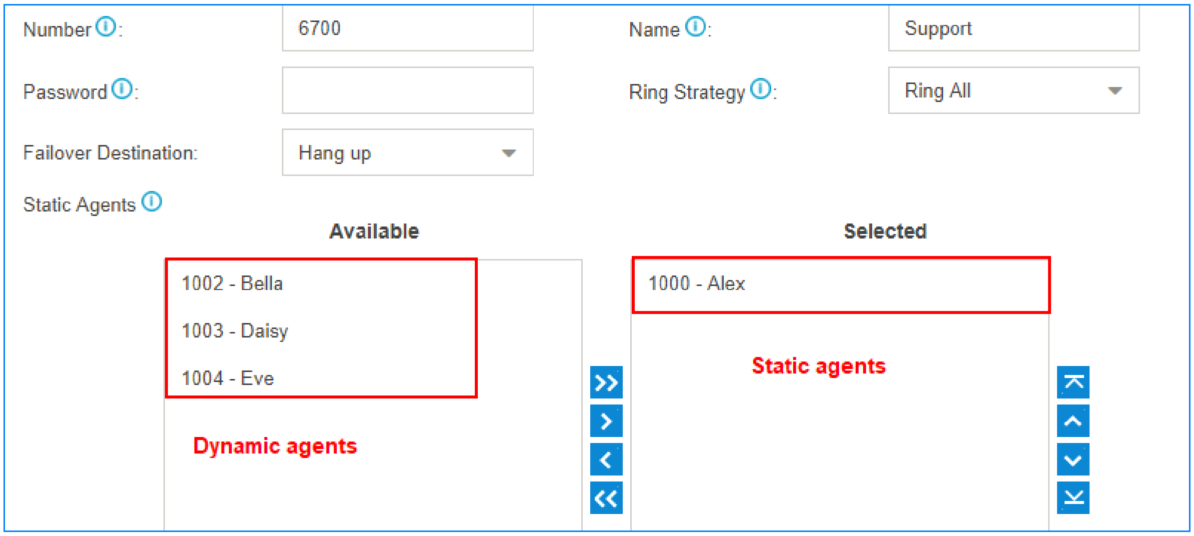
On the Queue configuration page, the unselected agents act as dynamic agents.



Add a Queue

Add a simple call queue.

1. Go to **Settings > PBX > Call Features > Queue**, click **Add**.
2. Specify a **Name** and **Number** for the queue.
3. Select a **Ring Strategy** for the call.
   - **Ringing All**: Ring All available Agents simultaneously until one answer.
   - **Least Recent**: Ring the Agent which was least recently called.
   - **Fewest Calls**: Ring the Agent with the fewest completed calls.
   - **Random**: Ring a Random Agent.
   - **Rememory**: Round Robin with Memory,Remembers where it left off in the last ring pass.
   - **Linear**: Rings interfaces in the order speciclied in the configuration file.
4. Select **Failover Destination**, define what should happen if the call does not get answered by an agent.
5. Select **Static Agents** for the queue.



- **Dynamic agents**: A dynamic agent can log in or log out a queue at any time.
- **Static agents**: A static agents will always stay in the queue.

6. Set the **Agent Timeout**, define how long the phone should keep ringing before it considers the call unanswered by that agent.
7. Click **Save** and **Apply**.

It is done for a simple call queue, for more information of queue settings, refer to Queue Settings.

Queue Settings

References of basic queue settings and caller experience settings.

Basic Queue Settings

Option	Description
Number	Use this number to dial into the queue, or transfer callers to this number to put them into the queue.
Name	Give this queue a brief name to help you identify it.
Password	You can require agents to enter a password before they can login to this queue.

Option	Description
Ring Strategy	This option sets the Ringing Strategy for this Queue. • **Ringing All**: Ring All available Agents simultaneously until one answer. • **Least Recent**: Ring the Agent which was least recently called. • **Fewest Calls**: Ring the Agent with the fewest completed calls. • **Random**: Ring a Random Agent. • **Rememory**: Round Robin with Memory,Remembers where it left off in the last ring pass. • **Linear**: Rings interfaces in the order specicied in the configuration file.
Failover Destination	Set the failover destination.
Static Agents	Select static agent of the queue. The static agents will always stay in the queue. **Note:** • The static agent is not allowed to log in and log out the queue. • The unselected users are dynamic agents.
Agent Timeout	The number of seconds an agent's phone can ring before we consider it a timeout. If you wish to customize, enter the value in the text box directly.
Ring In Use	If set to `no`, unchecked, the queue will avoid sending calls to members whose device are known to be “in use”.
Agent Announcement	Announcement played to the Agent prior to bridging in the caller.
Retry	The number of seconds to wait before trying all the phones again. If you wish to customize, enter the value in the text box directly.
Wrap-up Time	How many seconds after the completion of a call an Agent will have before the Queue can ring them with a new call .If you wish to customize, enter the value in the text box directly. Input 0 for no delay.

Call Experience Settings

Caller Settings	
Music On Hold	Select the “Music on Hold” playlist for this Queue.
Caller Max Wait Time	Select the maximum number of seconds a caller can wait in a queue before being pulled out. If you wish to customize, enter the value in the text box directly. Input 0 for unlimited.
Leave When Empty	If enabled, callers already on hold will be forced out of a queue when no agents available.
Join Empty	If enabled, callers can join a queue that has no agents.
Join Announcement	Announcement played to callers once prior to joining the queue.

Caller Settings	
Agent ID Announcement	Announcement played to the callers to prompt the agent ID. The agent is who will answer the call. • [None]: The system will not announce the agent ID. • [Default]: The system will play the prompt "{extension number} will be connected. Please wait". The {extension number} is the extension number of the agent. • Custom Prompt: If you choose your custom prompt. The system will play "{extension number}" + your custom prompt.
Satisfaction Survey Prompt	When the agent hangs up, the system will play the prompt to ask the caller to rate their satisfaction scale.
Caller Position Announcements	
Announce Position	Announce position of caller in the queue.
Announce Hold Time	Enabling this option causes PBX to announce the hold time to the caller periodically based on the frequency timer. Either yes or no; hold time will be announced after one minute.
Frequency	How often to announce queue position and estimated hold time.
Periodic Announcements	
Prompt	Select a prompt file to play periodically.
Frequency	How often to play the periodic announcements.
Events	
Key	Once the events settings are configured, the callers are able to press the key to enter the destination you set. Usually, a prompt should be set on **Periodic Announcements** to guide the callers to press the key.

Log in/out a Queue

A dynamic agent can log in or log out a queue at any time.

Log in/out a Queue by Feature Code

Note: If the static agents try to log out a queue, the system will play a prompt "Agent logged out, goodbye"; But actually, the agent is still in the queue.

- To log in a queue, dial `[QUEUE_NUM]*`.

  For example, dynamic agent 1000 dials `6700*` to log in the queue 6700.
- To log out a queue, dial `[QUEUE_NUM]**`.

  For example, dynamic agent 1000 dials `6700**` to log out the queue 6700.

- Dial `*75[QUEUE_NUM]` to log in a queue.

  For example, dynamic agent dials `*756700` to log in the queue 6700.
- Dial `*75[QUEUE_NUM]` again to log out a queue.

  For example, dynamic agent dials `*756700` again to log out the queue 6700.

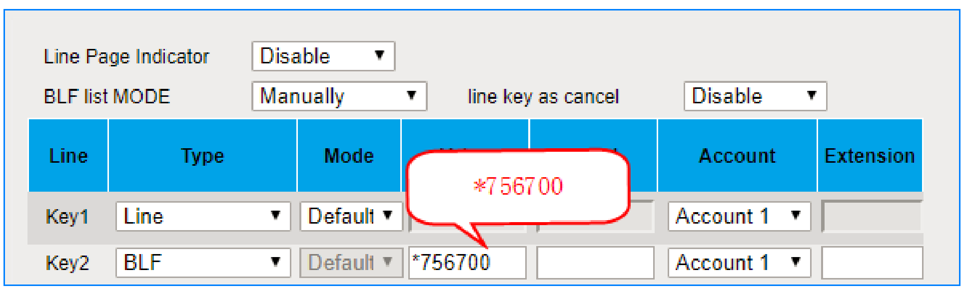
Log in/out a Queue by BLF Key

A dynamic agent can set a BLF key on his/her IP phone to quickly log in or log out a queue.

For example, on the phone of a dynamic agent, set a BLF key to quickly log in or log out queue 6700.

The following instructions are based on the Htek UC912 v2.0.4.4.33.

1. Log in the phone web interface, go to **Function Keys > Line Key**.
2. Set a BLF key to log in or log out queue 6700.



   - **Type**: Set to **BLF**.
   - **Value**: The BLF key format is `*75[QUEUE_NUM]`. In this example, set to `*756700`.
   - **Account**: Select the account that is registered to the extension number of the agent.
3. Click **SaveSet**.

Now, the agent can press the BLF key to switch his/her status in the queue.

- When the prompt "agent logged out,goodbye." is played, the agent is logged out of the queue.
- When the prompt "agent logged in, goodbye." is played, the agent is logged in the queue.

Monitor Agent Status by BLF

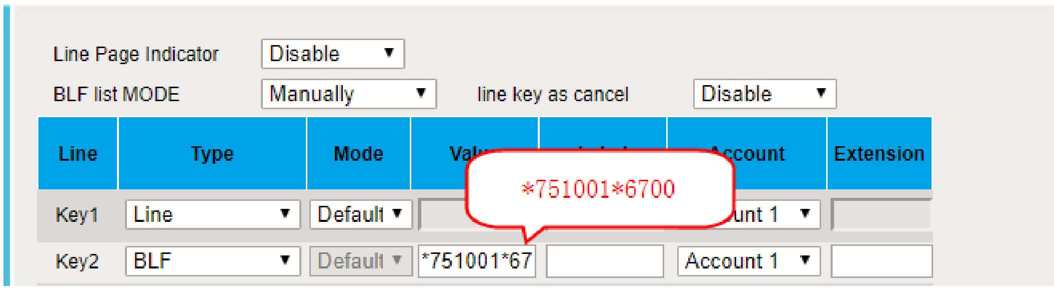
In a call center scenario, a supervisor can set BLF keys to monitor if the agents are in a specific queue. An agent can also set a BLF key to monitor his own status.

This topic is based on the Htek UC912 v2.0.4.4.33.

Note: Monitoring agent status is supported in the firmware version 30.8.0.8 or later.

We will set a BLF key to monitor if the agent 1001 is in the queue 6700 or not.

1. Log in the phone web interface, go to **Function Keys > Line Key**.
2. Set a BLF key to monitor extension 1001.



   - **Type**: Set to **BLF**.
   - **Value**: The BLF key format is `*75[EXT_NUM]*[QUEUE_NUM]`. In this example, set to `*751001*6700`.
   - **Account**: Select the account that has an extension registered to the PBX.

3. Click **SaveSet**.
   Check the BLF LED status:

   Note: Different brands of IP phone may have different LED indications.

   - **Green LED**: The agent 1001 is not in the queue 6700.
   - **Red LED**: The agent 1001 is in the queue 6700.
   - **BLF LED is off**: Check if your configurations are correct.

Conference

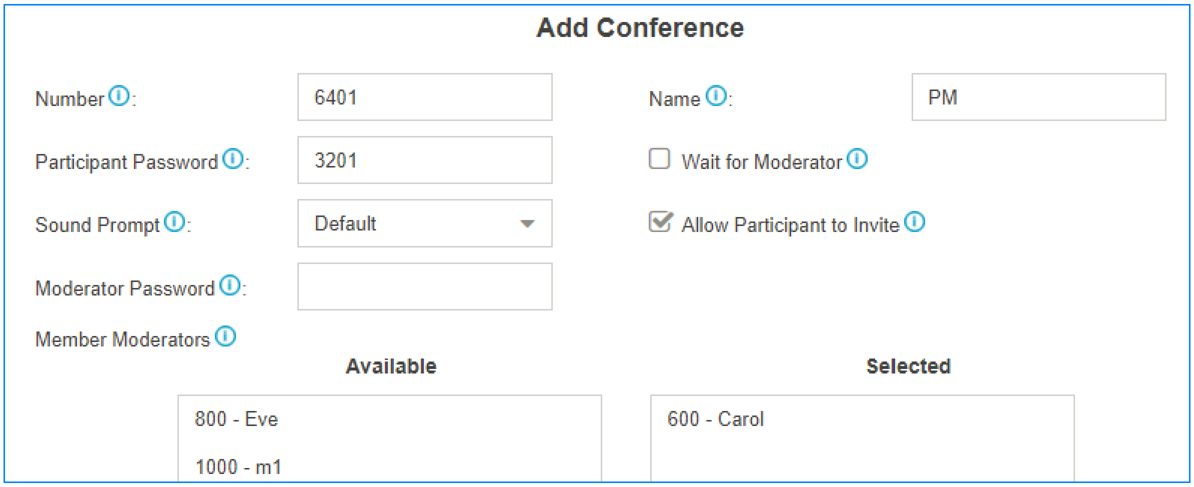
Conference calls increase employee efficiency and productivity, and provide a more cost-effective way to hold meetings.

Conference members can dial * to access to the settings options and the admin can kick the last user out and lock the conference room.

Add a Conference

To make a conference call, you should add a conference on the PBX first.

1. Go to **Settings > PBX > Call Features > Conference**, click **Add**.
2. On the configuration page, configure the Conference.



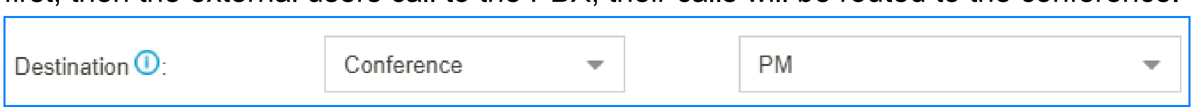
- **Number**: The extension users need to dial this number to join the conference.
- **Name**: Set a name for the conference.
- **Participant Password**: Optional. If the password is set, users need to input the correct PIN to join this conference.
- **Wait for Moderator**: If this option is checked, the conference participants could not hear each other until the moderator joins in the conference.
- **Sound Prompt**: Select the sound prompt used for the login and logout of conference members.
- **Allow Participant to Invite**: Whether to allow the participants to invite users to join the conference.
- **Moderator Password**: The moderator doesn't need to enter a password to join the conference. If a user enter this password to join the conference, he/she will act as the conference moderator.
- **Member Moderators**: Select the conference moderators.

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

Join a Conference

Both the PBX extension users and the external users can join the conference.

1. For the PBX extension users, dial the conference number to join the conference room.
2. For the external users, you need to set the inbound route destination to a conference first, then the external users call to the PBX, their calls will be routed to the conference.



Conference Voice Menu

During the conference call, the users could manage the conference by pressing * key on their phones to access voice menu for conference room.

Conference Moderator Voice Menu	
1	Mute/ un-mute yourself.
2	Lock /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.
Conference Users IVR Menu	
1	Mute/ un-mute yourself.
4	Decrease the conference volume.
6	Increase the conference volume.
7	Decrease your volume.
8	Exit the voice menu.
9	Increase your volume.

Call Pickup

Call Pickup is a feature that allows a user to answer an incoming call that rings on a telephone other than the user's own.

Extension Call Pickup

When a user wants to pick up a call that is ringing at the other extension that is not in the same pickup group, the user can dial "Extension Pickup feature code (default *04) + Extension Number" to pick up the call.

Extension Call Pickup Feature Code

The default Extension Call Pickup feature code is *04.

You can change the code on **Settings > PBX > General > Feature Code > Extension Pickup**.

Operation

Dial *04[EXT_NUM] to pick up a call.

For example, the ringing extension number is 1000, you should dial *041000 to pick up the call.

Pick up an Extension's Call by BLF

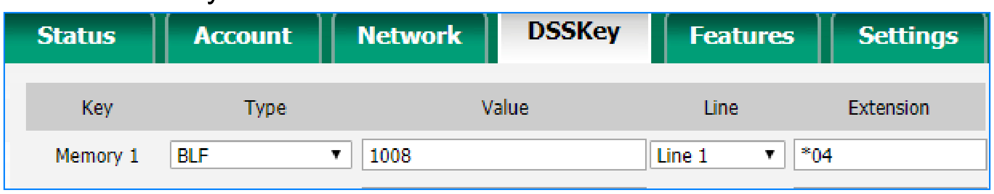
You can set a BLF key of Extension Call Pickup on your phone. The BLF key will show the real-time status of the extension. When the extension is ringing, you can press the BLF key to pick up the call.

We take Yealink T27G v69.82.0.20 as an example below.

1. Set a BLF key to monitor and pick up an extension.

    a. Log in the phone web interface, go to **Dsskey** page.

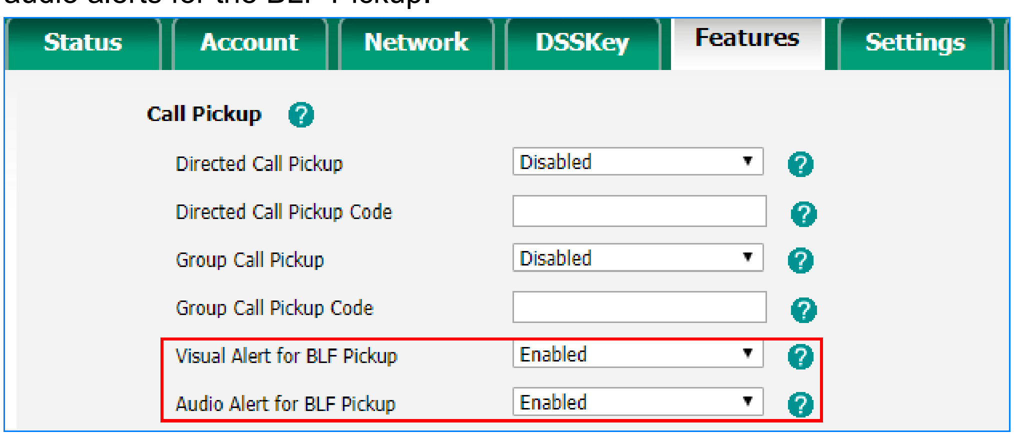
    b. Set the BLF key as below.



    - **Type**: Select **BLF**.
    - **Value**: Enter the extension number that you want to monitor.
    - **Line**: Choose the line where your extension is registered on.
    - **Extension**: Enter the feature code of extension pickup. The default code is *04.

    c. Click **Confirm**.

2. To get notified when the monitored extension has an incoming call, set visual alerts and audio alerts for the BLF Pickup.



    a. On the phone web page, go to **Phone > Features > Call Pickup**.

b. In the **Visual Alert for BLF Pickup**, select **Enabled**.
When a call reaches the monitored extension, you can see the incoming caller ID on your phone.

c. In the **Audio Alert for BLF Pickup**, select **Enabled**.

A “beep” sound will remind you of an incoming call for the monitored extension.

d. Click **Confirm**.

If your configuration is correct, the BLF LED will turn green.

When the monitored extension has an incoming call, the followings occur on your phone, press BLF key to pick up the call.

- The phone plays a warning tone.
- The BLF LED turns red.

Group Call Pickup

If extension users are in the same pickup group, they can dial the Group Call Pickup feature code (default *4) to pick up the group member's incoming call.

Group Call Pickup Feature Code

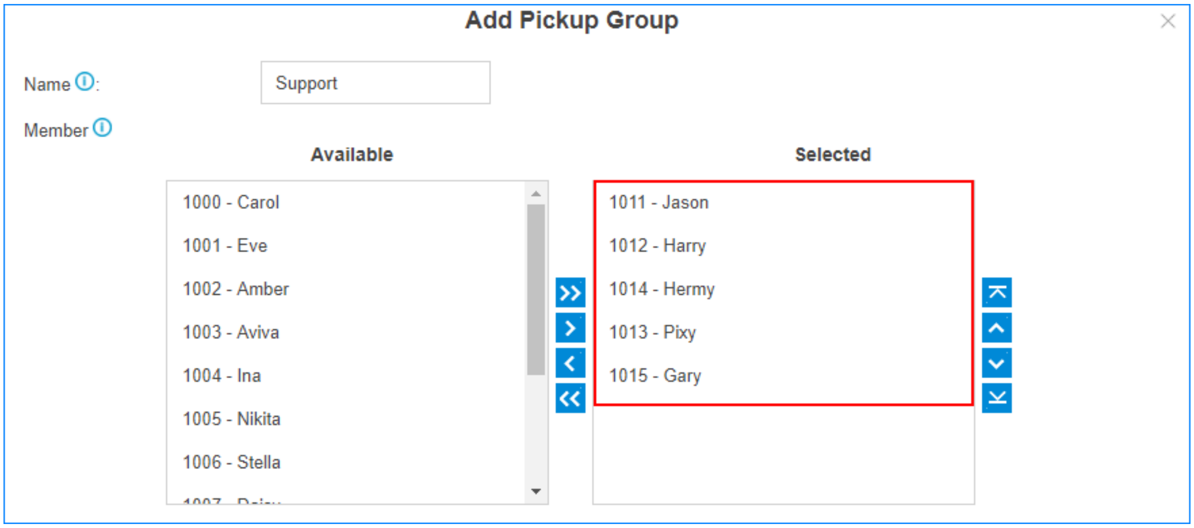
The default Group Pickup feature code is *4.

You can change the code on **Settings > PBX > General > Feature Code > Call Pickup**.

Add a Pickup Group

Generally, You can set the extension users who are in the same department in a pickup group.

1. Go to **Settings > PBX > Call Features > Pickup Group**, click **Add**.
2. Set the pickup group.



- **Name**: Give the group a name to help you identify it.
- **Member**: Select the desired extensions from **Available** box to **Selected** box.

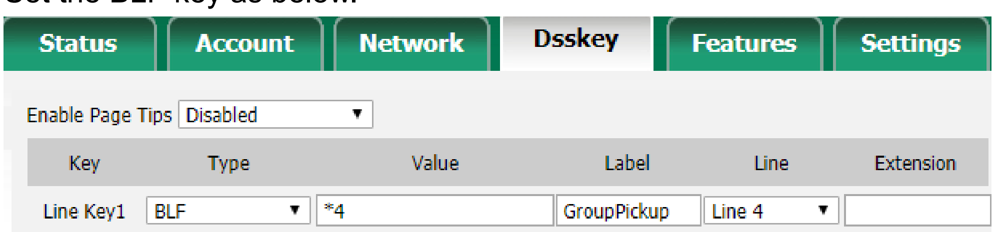
3. Click **Save** and **Apply**.

Pick up A Group Member's Call by BLF

You can set a BLF key for Group Call Pickup on your IP phone. When your group member's phone is ringing, you can press the BLF key to quickly pick up the call.

We take Yealink T27G v69.82.0.20 as an example below.

1. Log in the phone web interface, go to **Dsskey** page.
2. Set the BLF key as below.



- **Type**: BLF
- **Value**: Enter the feature code of group pickup. The default code is *4.
- **Label**: Set a label that you want to display on the phone screen.
- **Line**: Choose the line where your extension is registered on.

3. Click **Confirm**.
   If your configuration is correct, the BLF LED will turn green.

Call Transfer

Yeastar Cloud PBX supports Attended Transfer and Blind Transfer, users can dial the feature code to transfer a call on their phones.

Attended Transfer (Default feature code *3)

An attended transfer, also called consult transfer or warm transfer, is when you speak with the new person before the call is transferred. You can tell the new person about the caller's issue and give any background information before transferring the call (without the caller hearing).

Blind Transfer (Default feature code *03)

A blind transfer is when you transfer the caller to another person without speaking to the new person first.

Attended Transfer

If you want to tell the new person about the caller's issue and give any background information before transferring the call, you can choose attended transfer.

Scenario: You (B) are talking with A, then transfer the call to C.

1. During the call with person A, dial *3 on your phone.
   You will hear the prompt "transfer" and the dial tone.
2. Dial C's number.
   C's phone is ringing. After C answers the call, the call between you and C is established. In this time, the call between you and A is held.
3. Hang up your call, the call between A and C is established.

Blind Transfer

If you don't need to consult the new person who you want to transfer the call to, you can perform a blind transfer. Your call will be ended after you transfer the call.

Scenario: You (B) are talking with A, then transfer the call the C.

1. During the call with person A, dial *03 on your phone.
   You will hear the prompt "transfer" and the dial tone.
2. Dial C's number and hang up.
   C's phone is ringing. After C answers the call, the call between A and C is established.

Callback

Callback feature allows callers to hang up and get called back to the PBX. Callback feature could reduce the cost for the users who work out of the office using their own mobile phones.

Set up Callback

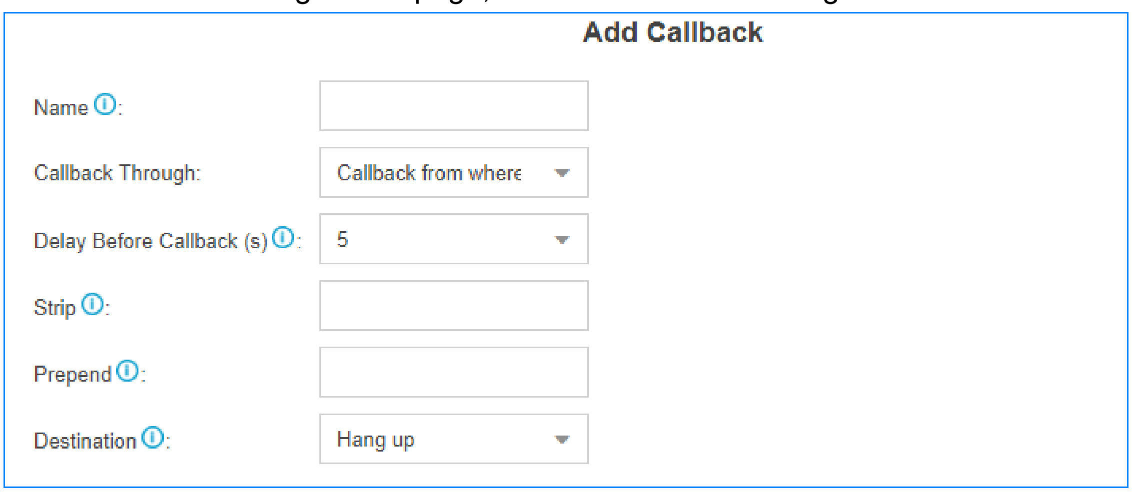
Add a Callback rule and set Inbound Route destination to the Callback rule.

Note: Make sure that the Caller ID service is enabled on the callback trunk. If the PBX cannot recognize the inbound caller ID, callback will fail.

1. Add a Callback rule.

    a. Go to **Settings > PBX > Call Features > Callback**, click **Add**.

    b. On the Callback configuration page, finish the callback settings.



- **Name**: Set a name for the Callback.
- **Callback Through**: Select which trunk to use when calling back.

  Note: Make sure that you have set up an outbound route for the trunk, or callback will fail. If the Register-Trunk is used for Callback, make sure the **From User** is configured, or callback would fail.

- **Delay Before Callback**: How long to wait before calling back the caller.
- **Strip**: Optional. How many digits will be stripped from the call in number before the callback is placed.

  Note: You do not need to configure **Strip** if the trunk supports calling back with the Caller ID directly.

For example, user 5503301 calls in the PBX, the caller ID displays 05503301. To call back to the user, you should set strip 1 digit so that the PBX will call back to 5503301.

- **Prepend**: Optional. The digits added before a callback number before the callback is placed.

  Note: You do not need to configure **Prepend** if the trunk supports calling back with the Caller ID directly.

  For example, user 15880232154 calls in the PBX, the caller ID displays 15880232154. To call back to the long-distance number 15880232154 through the selected trunk, you should add digit 9 before the number. In this case, set **Prepend** to **9**.

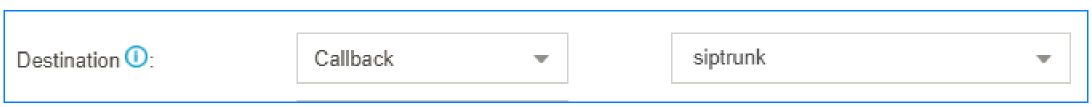
- **Destination**: Where the callback will direct the caller to.

c. Click **Save** and **Apply**.

2. Set Inbound Route destination to callback.

   a. Go to **Settings > PBX > Call Control > Inbound Route**, edit your inbound route.

   b. Set the Inbound **Destination** to the Callback.



   c. Click **Save** and **Apply**.

3. Test callback.
   Make an inbound call to the PBX trunk, after you hear the ring tone, hangup the call, the PBX will call back to you.

Speed Dial

Sometimes you may just need to call someone quickly without having to look up his/her phone number. You can by simply define a shortcut number. You can use Speed Dial feature to place a call by pressing a reduced number of keys.

Add a Speed Dial Number

1. Go to **Settings > PBX > Call Features > Speed Dial**, click **Add**.
2. On the configuration page, configure the Speed Dial.

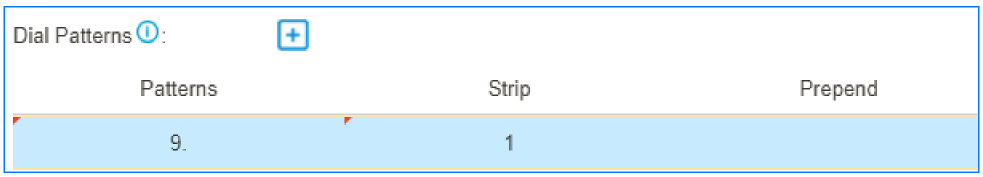
- **Speed Dial Code**: Speed dialing number.
- **Phone Number**: The phone number that you want to call.

Note: You need to add the outbound dial prefix before the phone number if you want to call an external number.

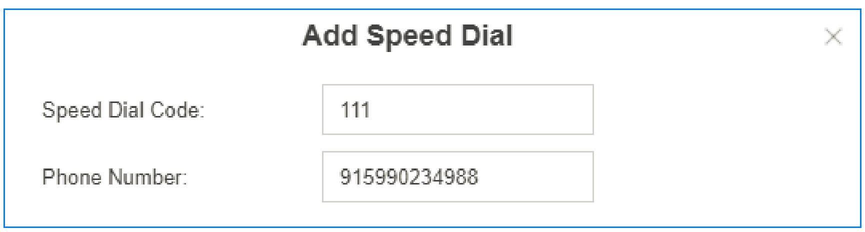
3. Click **Save** and **Apply**.

Speed Dial Example

Assume that you have an outbound route set as below, and you will dial speed number 111 to reach an external number 15990234988 through the route.



You need to set the Speed Dial as below:



Dial *99111 on your phone to call the number 15990234988. *99 is the default feature code for speed dial.

DISA

DISA (Direct Inward System Access) allows users outside the office to make calls through the PBX's trunks. For the staffs who are outside the office, they can use DISA feature to take advantage of lower long-distance rates that are provided by the PBX trunks.

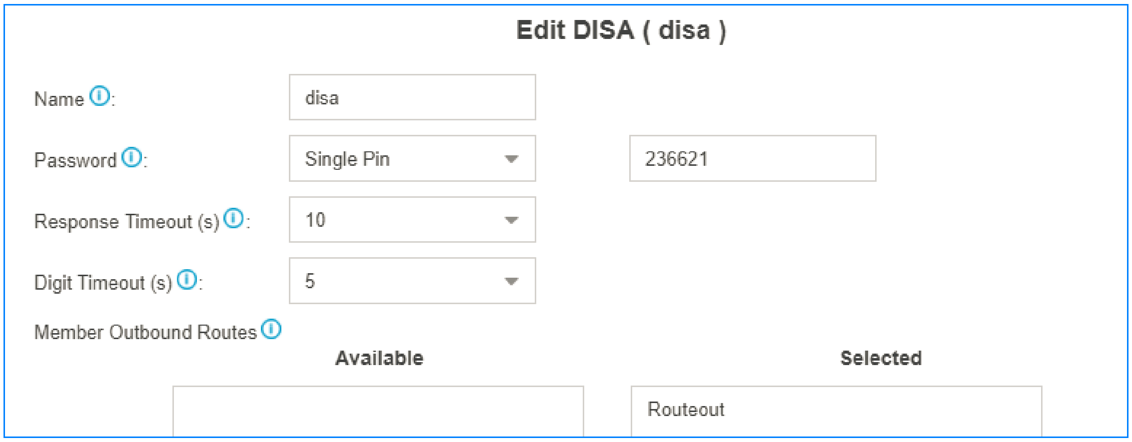
Set up DISA

Add a DISA and set the Inbound Route destination to DISA.

1. Add a DISA.

    a. Go to **Settings > PBX > Call Features > DISA**, click **Add**.

    b. On the DISA configuration page, finish the DISA configurations.



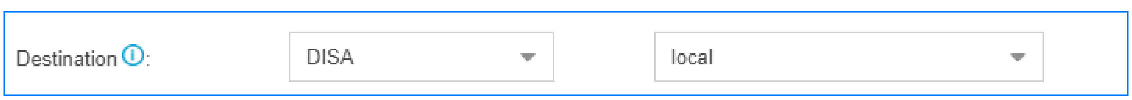
- **Name**: Set the DISA name.
- **Password**: Set password for the DISA.
- **Response Timeout**: The maximum amount of time it will wait before hanging up if the user has dialled an incomplete or invalid number.
- **Digit Timeout**: The maximum amount of time permitted between digits.
- **Member Outbound Routes**: Select the outbound routes that can be accessed from the DISA.

c. Click **Save** and **Apply**.

2. Set Inbound Route destination to DISA.

   a. Go to **Settings > PBX > Call Control > Inbound Route**, edit your inbound route.

   b. Set the Inbound **Destination** to the DISA.



   c. Click **Save** and **Apply**.

3. Test DISA.
   a. Make an inbound call to the PBX, you will get a dial tone after inputting a correct DISA pin code.
   b. Dial the external number that you want to call.

Intercom/Paging

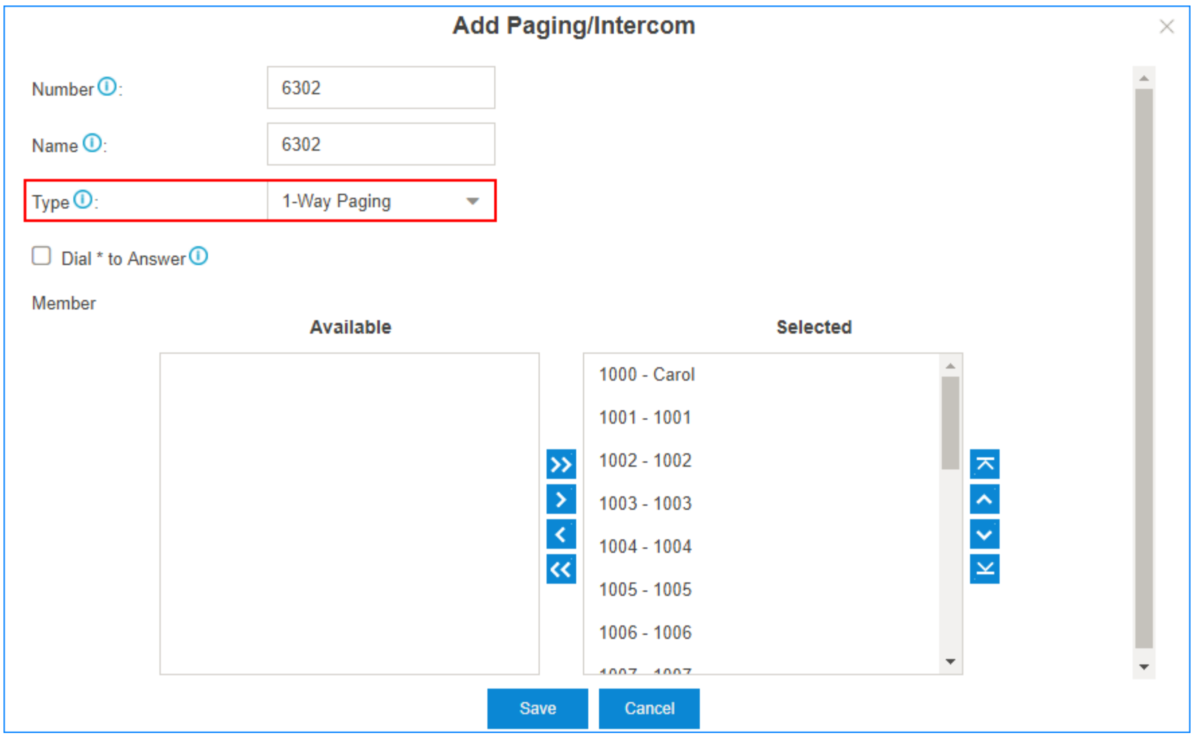
The Paging and Intercom features allow you to make an announcement to a group of extensions. The called parties do not need to pick up the handset as the audio will be played via the phone speakers.

Set up 1-Way Paging

Paging is used to make an announcement over the speakerphone to a phone or group of phones. The called parties will not ring, but instead answer immediately into speakerphone mode.

Note: Paging is typically one way for announcements only.

1. Go to **Settings > PBX > Call Features > Paging/Intercom**, click **Add**.
2. Set a 1-Way paging group.



   - **Number**: Use the default or specify an number for the paging group.
   - **Name**: Enter a name for the paging group.
   - **Type**: Choose `1-Way Paging`.
   - **Dial * to Answer**: If this option is checked, the paging group members can dial `*` to talk to the paging initiator.

     Note: When a member dials `*`, the group announcement will terminate, and the member who dials `*` can have a private call with the paging initiator.

   - **Member**: Choose the group members to the **Selected** box.
3. Click **Save** and **Apply**.

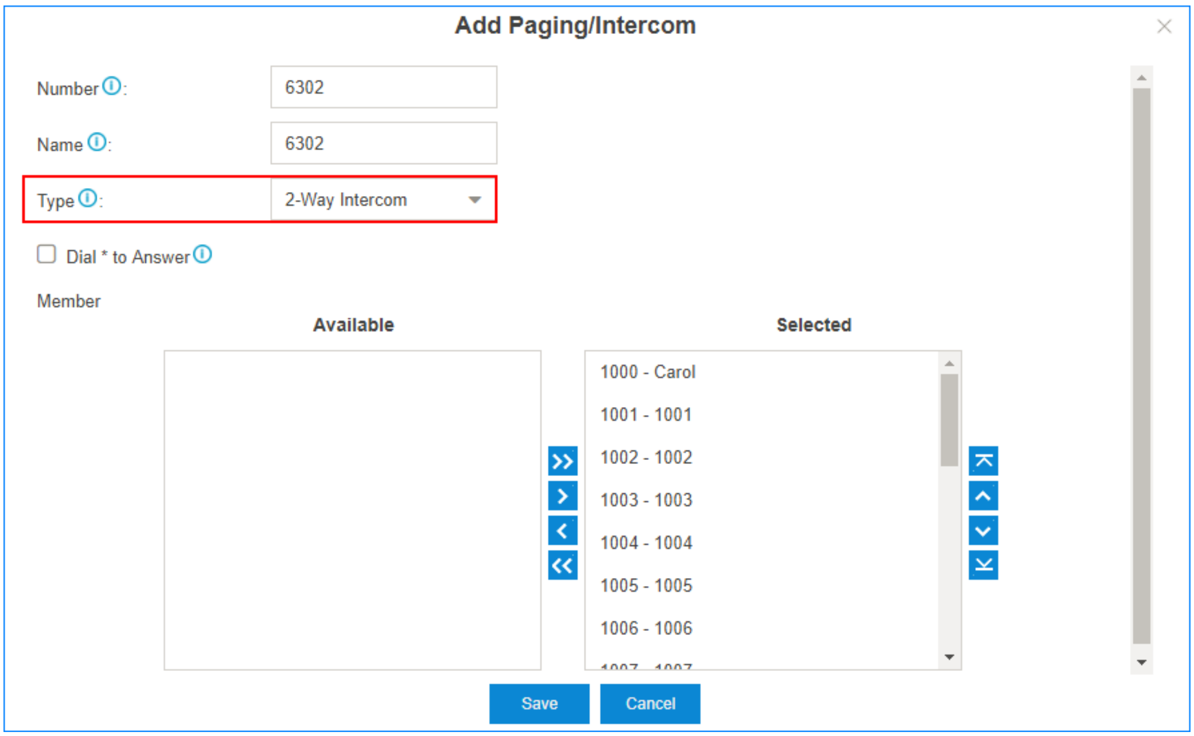
When you dial the paging group number, the members in the group will hear the announcement.

Set up 2-Way Intercom

2-way intercom is used to make a multi-party conference. The called parties will automatically answer the call into speakerphone mode and join the conference.

Note: Intercom allows all users in the group to talk and be heard by all.

1. Go to **Settings > PBX > Call Features > Paging/Intercom**, click **Add**.
2. Set a 2-Way intercom group.



- **Number**: Use the default or specify an number for the intercom group.
- **Name**: Enter a name for the intercom group.
- **Type**: Choose `2-Way Intercom`.
- **Dial * to Answer**: If this option is checked, the intercom group members can dial `*` to talk to the intercom initiator.

  Note: When a member dials `*`, the group announcement will terminate, and the member who dials `*` can have a private call with the intercom initiator.

- **Member**: Choose the group members to the **Selected** box.

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

When you dial the intercom group number, the members in the group will automatically join the conference by speakerphone mode.

Set up 1-Way Multicast Paging

Multicast Paging allows you to easily and quickly broadcast instant audio announcements to phone users who are listening to the same multicast IP address of the PBX.

When you make a Multicast Paging, the PBX sends Real-time Transport Protocol (RTP) streams to the IP phones without involving SIP signaling. The phones that will receive the RTP streams don't need to register SIP extensions.

Note:

- The IP phone that will receive 1-way multicast paging should support Multicast Paging feature.
- The Multicast Paging is one-way audio call.

1. Set a 1-way Multicast Paging on the PBX.

   a. Go to **Settings > PBX > Call Features > Paging/Intercom**, click **Add**.

   b. Set a 1-Way multicast paging.

   - **Number**: Use the default or specify an number for the paging group.
   - **Name**: Enter a name for the paging group.
   - **Type**: Choose `1-Way Multicast Paging`.
   - **IP of Multicast Channel**: Enter the multicast IP address and port (e.g. 224.255.255.255:1000).

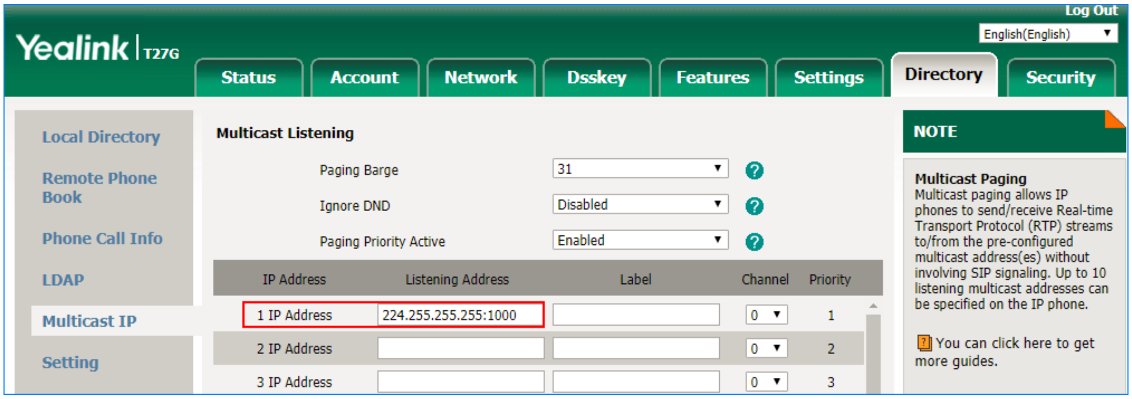
     Note: The range of multicast IP address is 224.0.0.0 - 239.255.255.255.

   c. Click **Save** and **Apply**.

2. Set Multicast Paging on each of your IP phone.
   In the following, we take Yealink T27G as an example.

a. Log in the phone web interface, go to **Directory > Multicast IP**.

b. In the **Multicast Listening** section, enter the same multicast IP address and port of the PBX.



c. Click **Confirm**.

When you dial the paging group number, the members in the group will automatically answer the call into speakerphone mode.

Note: If the multicast paging doesn't work, check the following:

- Multicast IP address and port are in the correct range.
- If the PBX and IP phones are in different IP segments (e.g. PBX is in `192.168.5.X` IP segment, IP phones are in `192.168.3.X` IP segment), check if your router supports IP Multicast in different IP segments.

Make an Announcement to a Specific User

Extension users can dial the intercom feature code to make an intercom to a specific extension, the called party can respond immediately without picking up the handset.

The default Intercom feature code is *5.

Note: In this way, the audio is two way, both the caller and called party can hear each other.

Extension user 2000 makes an intercom call to extension user 1000.

1. Dial `*51000` on the phone of extension 2000.

   The call on extension 1000 will be answered automatically.

Call Parking

Call Parking is a feature that allows you to suspend a call for an extended period of time and then retrieve that call from any extension.

Scenarios

During a call with clients, extension users may need to check information somewhere else. In such case, extension users can park the call temporarily and retrieve the call by any extensions when getting things done.

Settings of Call Parking

Go to **Settings > PBX > General > Feature Code > Call Parking**, you can modify the feature code, parking extension range, and parking time.

We provide default settings of call parking as follows.

Settings	Descriptions
Call Parking	The default feature code is *6. During a call, dial *6 on your phone, the system will automatically assign a parking slot number to the call.
Directed Call Parking	The default feature code is *06. During a call, dial "06+parking slot number", the call will be parked to the designated parking slot number.
Parking Extension Range	Specify the range of parking extension where a call will be parked. The default value is 6900-6999. **Note:** The rang of parking extension must be different from existing extension ranges (**Settings > PBX > General > Preferences > Extension Preferences**).
Parking Timeout (s)	Specify the time that a call can be parked before it is retrieved by other extensions. The default value is 60s. **Note:** Parking Timeout must be longer than 30s.
Timeout Destination	If a parked call hasn't been retrieved before the parking timeout, PBX will route the call to the designated destination. • **Original Parker**:The call will be routed to the user who parks this call. • **Extension**: Te call will be routed to the designated extension number. • **Extension's Voicemail**: The call will be routed to the designated extension's voicemail. • **Custom Number**: The call will be routed to the designated number.

Call Parking (Default feature code: *6)

You can dial the feature code of Call Parking to get the parking slot number, then dial the parking slot number on another phone to retrieve the call.

Example:

1. During a call, dial *6 on your phone, the system will prompt you that the parking slot number is 6900.
2. Dial 6900 on another phone to retrieve the call.

Direct Call Parking (Default feature code: *06)

If you get a parking slot number from your administrator, you can dial the “feature code of Direct Call Parking+parking slot number" to park the call to the slot.

Example:

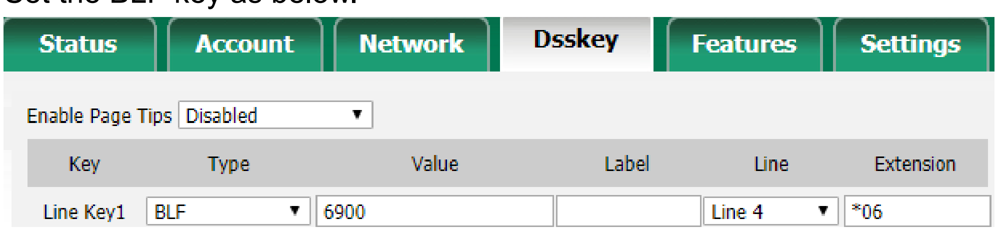
1. During a call, dial *066900 to park the call to slot 6900.
2. Dial 6900 on another phone to retrieve the call.

Park Calls by BLF

You can set a BLF key of Call Parking on your phone. The BLF key will show the real-time status of the parking slot. If the parking slot is vacant, you can press the BLF key to park a call to the parking slot.

We take Yealink T27G v69.82.0.20 as an example below.

1. Log in the phone web interface, go to **Dsskey** page.
2. Set the BLF key as below.



   - **Type**: Select **BLF**.
   - **Value**: Enter the parking slot number.
   - **Line**: Select the line where your extension is registered on.
   - **Extension**: Enter the feature code of Direct Call Parking. The default code is *06.

3. Click **Confirm**.

- When the parking slot is vacant, the BLF LED is green.

Press the BLF key to park a call to the parking slot.

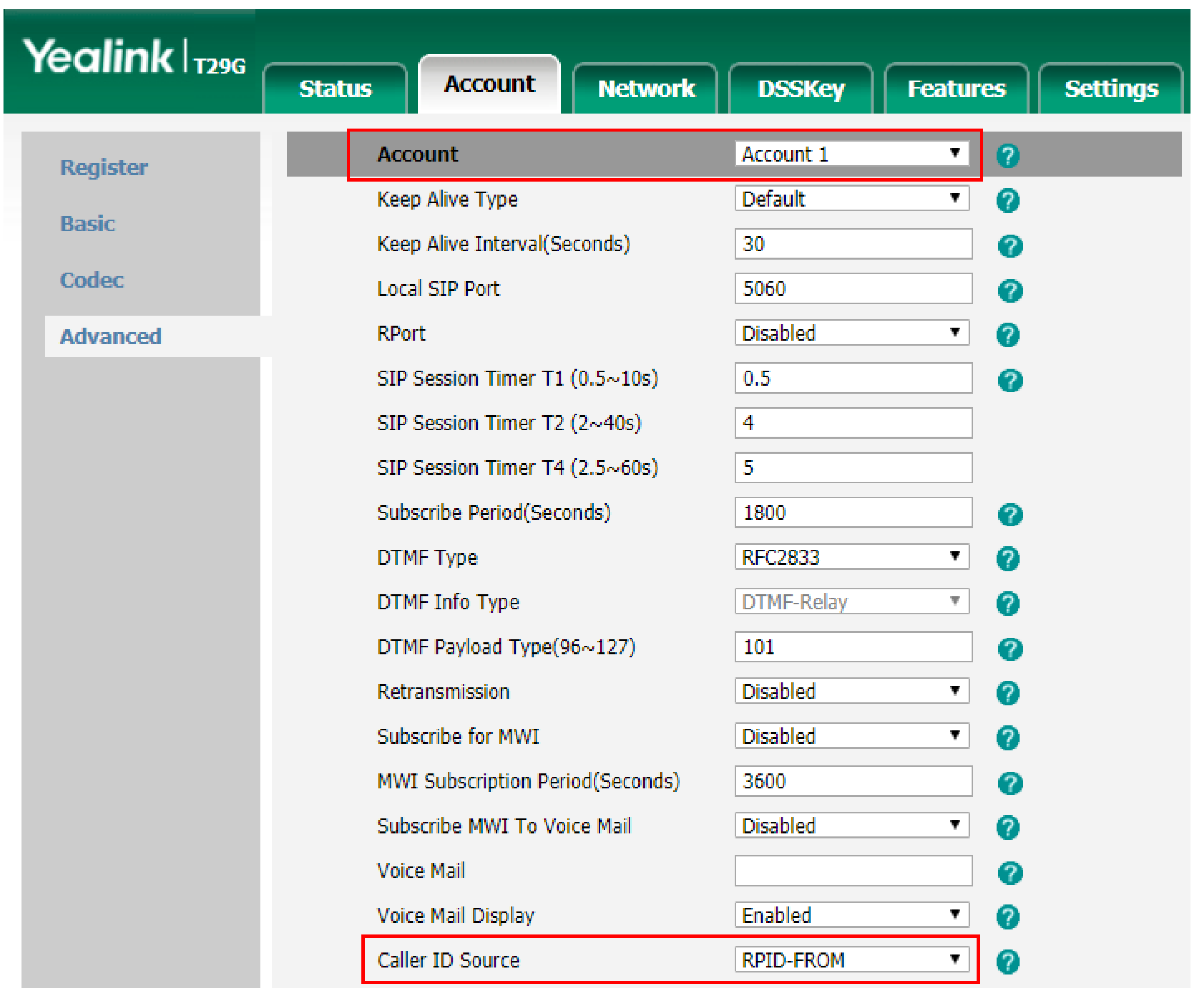
- When the parking slot is occupied, the BLF LED is red.

Configure Call Parking Caller ID

By default, when you retrieve a parked call, the call-park slot number (e.g. 6900) will be displayed on the phone. To display the original caller ID of the user who you were talking to, you need to configure SIP settings to get caller ID from Remote- Party-ID SIP header.

1. On PBX, enable **Send Remote Party ID**.
    a. Go to **Settings > PBX > General > SIP > Advanced**.
    b. Check the option **Send Remote Party ID** option.
    c. Click **Save** and **Apply**.
2. On the IP phone that you will use to retrieve a parked call, configure the **Caller ID Source**.

    Note: We take Yealink T29G v46.83.0.50 as an example below.

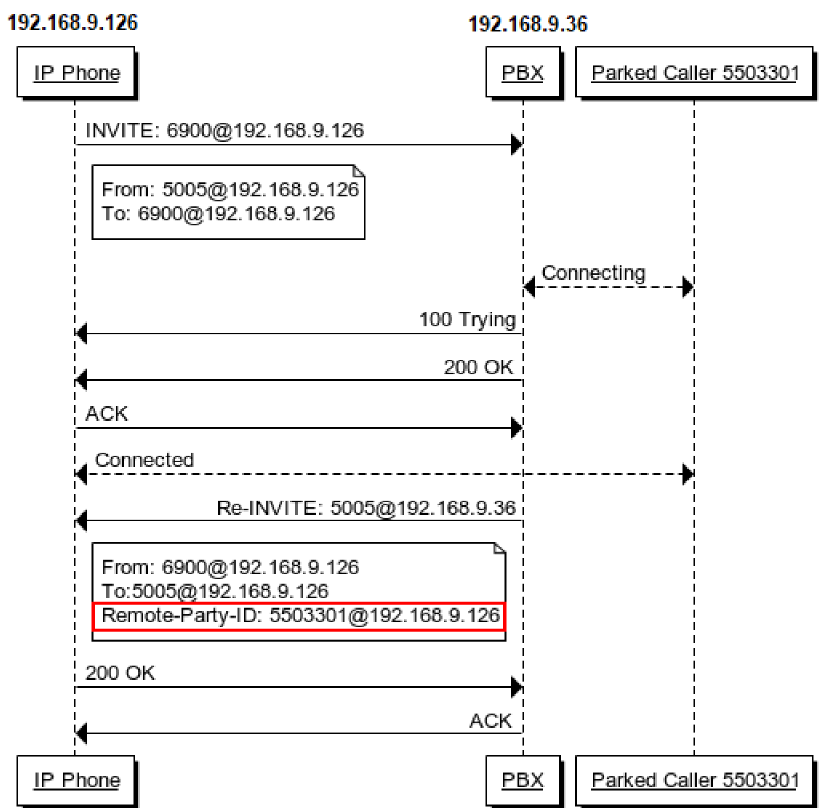


a. Log in the phone web interface, go to **Account > Advanced**.
b. In the **Account**, select the account where the extension is registered.
c. In the **Caller ID Source** field, select **RPID-FROM**.
d. Click **Confirm**.

Test call parking. When you retrieve the parked call from the IP phone, the phone screen will display the parking slot number for 1 or 2 seconds, then display the original caller ID.

The following call flow shows how the IP phone gets caller ID when a user retrieves a parked call.

1. A user dials parking slot number 6900 on IP phone to retrieve a parked call.
2. PBX sends a Re-INVITE packet that contains Remote-Party-ID.
3. The IP phone gets the caller ID from the Remote-Party-ID header.



Fax

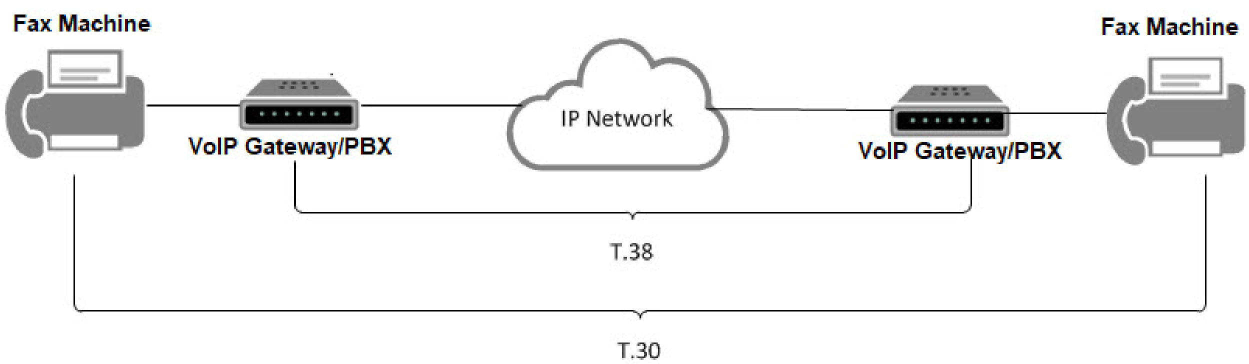
Yeastar Cloud PBX supports Fax over IP. You can send or receive a fax via a physical fax machine or receive a fax over the network.

What is T.38 Fax over IP?

T.38 is a protocol for sending faxes over a voice over IP (VoIP) network or the Internet in real time.

T.38 protocol defines the transport of data (a fax) between PSTN fax terminals through a fax gateway, between two Internet-aware fax terminals, or from a PSTN fax terminal through a fax gateway to an Internet-aware fax terminal. A T.38 stream is sometimes referred to as Fax over IP (FoIP).

PSTN fax terminals traditionally use the T.30 protocol to send analog data. To exchange analog fax data with a PSTN terminal over the Internet, the T.38 protocol first converts analog data into digital data. The protocol then converts the data back to analog on the receiving end if the receiver is a PSTN fax terminal.



T.38 Fax Settings

If the Fax over IP doesn't work, you can go to **Settings > PBX > General > SIP > T.38** to change the T.38 settings.

- **No Re-invite SDP Add T.38 Attribute**

  If this option is enabled, no T.38 attributes will be added in re-invite SDP packet.

- **Error Correction**

Error Correction Mode (ECM) for the Fax.

- **T.38 Max BitRate**

  T38 Max Bit Rate.

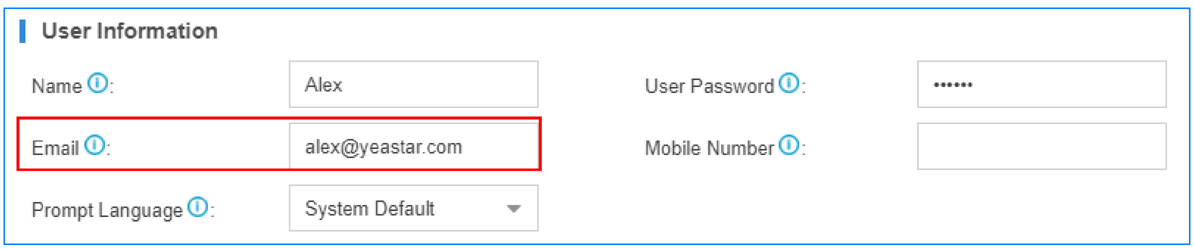
Fax to Email

Fax to Email feature helps you receive faxes on your smart phone or computer. Yeastar Cloud PBX will convert the received fax and forward it to an extension user's email.

Steps to Configure 'Fax to Email'

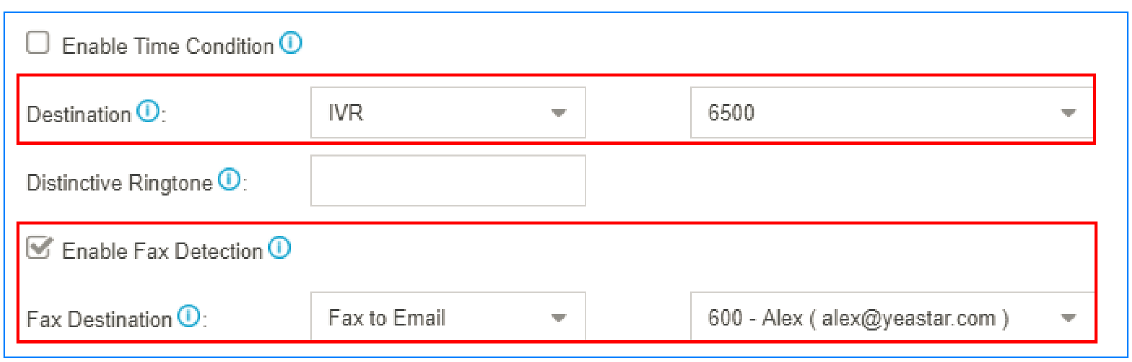
1. Configure the PBX **System Email**.

   Make sure the PBX system email works, or the PBX cannot forward the received faxes to an extension user's email.
2. Check if the extension user's email is configured.

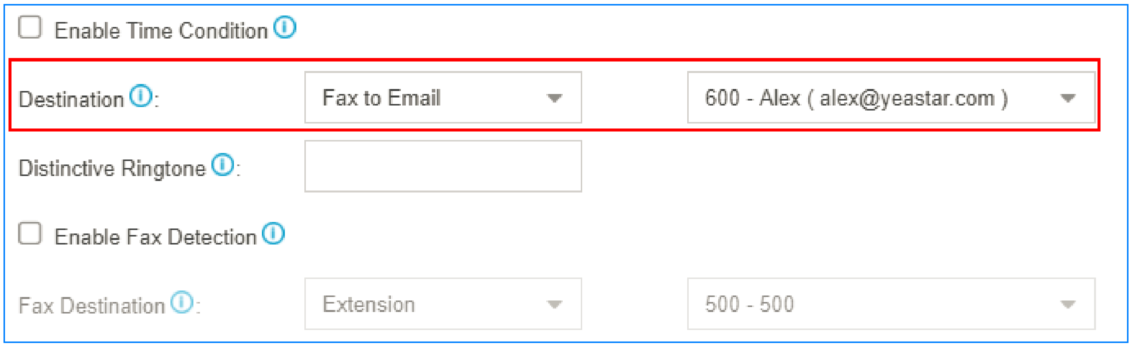


3. Configure the destination of your inbound route.

   - If you want to receive fax via fax detection, set the **Destination** to `IVR`, and set **Fax Destination** to `Fax to Email`.



   - If you want to receive fax through a private trunk, set the **Destination** to `Fax to Email`.



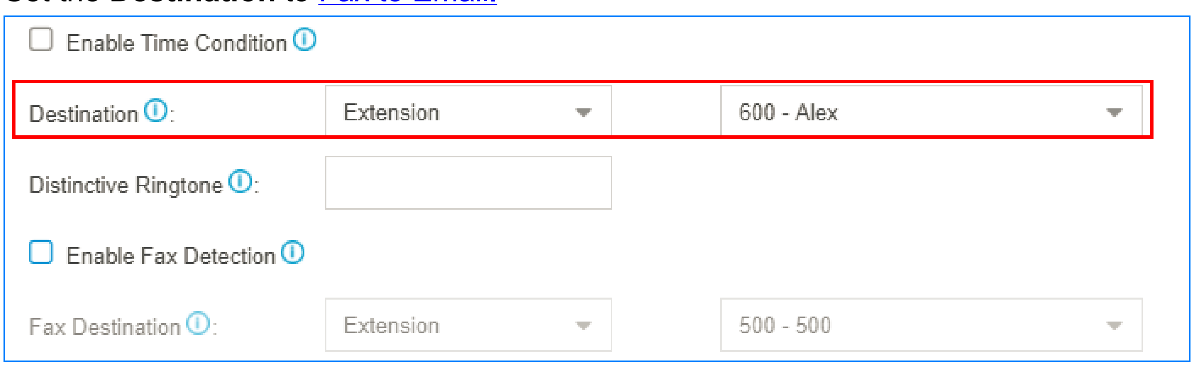
Receive Fax through a Dedicated Trunk

You can assign one or more trunks to receive faxes, and tell your customers to send faxes to the dedicated trunk number.

1. Go to **Settings > PBX > Call Control > Inbound Route**, click **Add**.
2. On the configuration page, select the dedicated trunk to the **Selected** box.

   Note: The selected trunk will be used to receive faxes only, users cannot make audio calls through the selected trunk.

3. Set the **Destination** to [Fax to Email.](#)



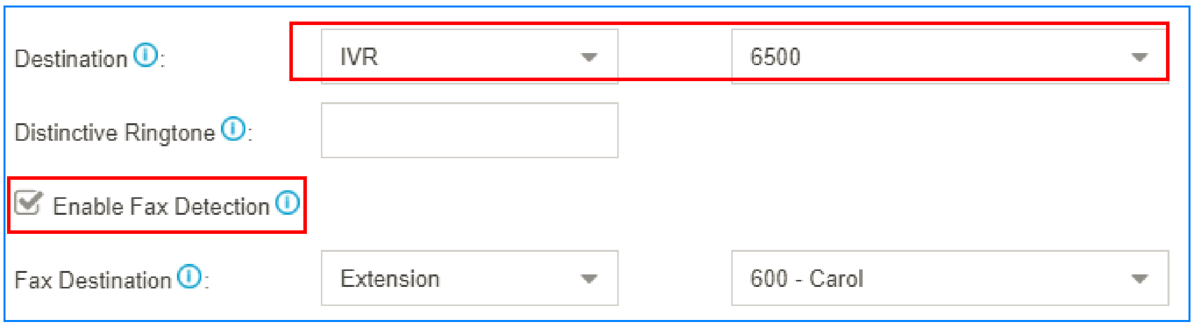
4. Click **Save** and **Apply**.

Users can dial the number of the dedicated trunk, then send fax to the PBX.

Receive Fax via Fax Detection

If you want to receive calls and also receive faxes through a trunk, you can set fax detection on your inbound route.

1. Go to **Settings > PBX > Call Control > Inbound Route**, configure your inbound route.
2. Select the trunk to the **Selected** box.
3. Set the **Destination** to `IVR`.
4. Check the option **Enable Fax Detection**.
5. Set the **Fax Destination** to [Fax to Email].

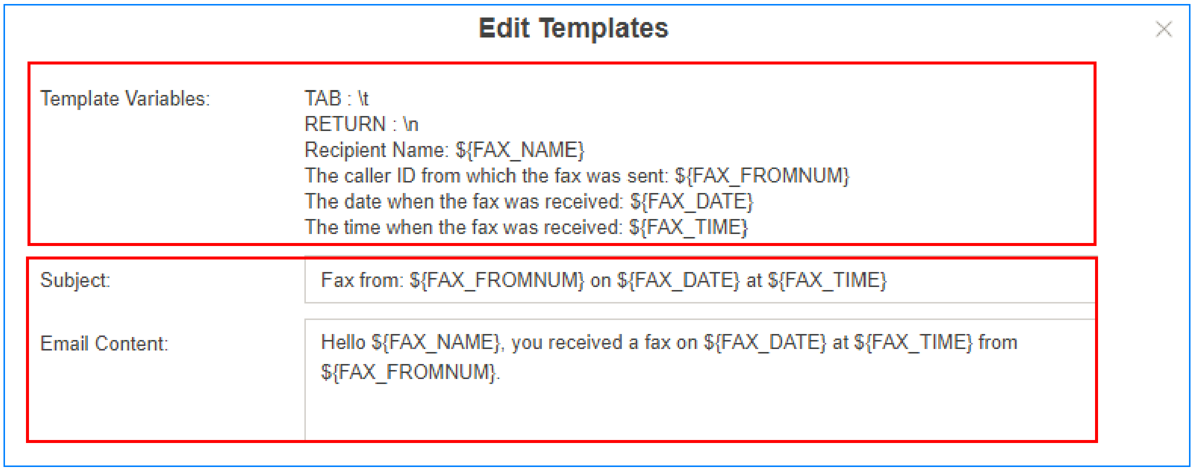


6. Click **Save** and **Apply**.

Edit 'Fax to Email' Template

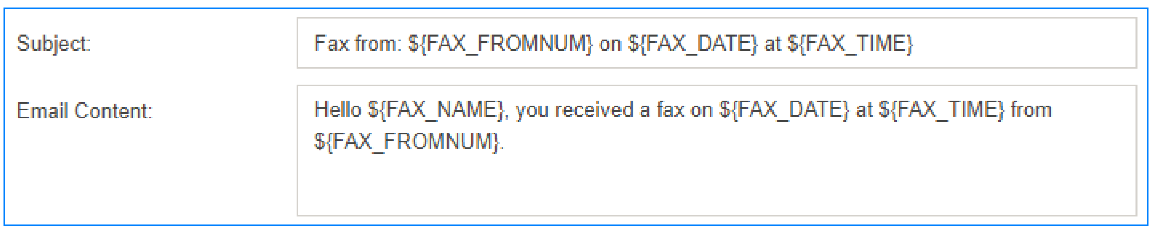
The PBX has a default email template for **Fax to Email**. You can edit the template according to your needs.

1. Go to **Settings > System > Email > Email Template**, click beside **Fax to Email**. On the **Edit Template** page, the description of variables and the default email contents are displayed.



2. Edit the email subject and email contents.

   Note: The variable names are unchangeable.



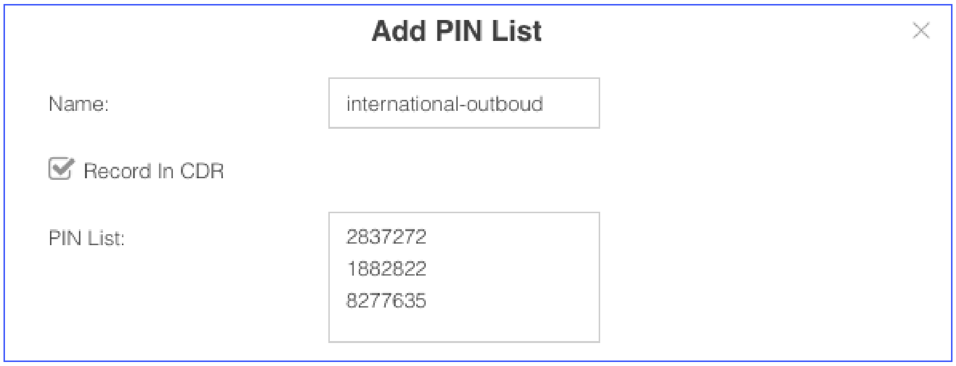
3. Click **Save** and **Apply**.

PIN List

PIN List is used to manage lists of PINs (numerical passwords) that can be used to access restricted features such as outbound route and DISA.

Add a PIN list

1. Go to **Settings > PBX > Call Features**, click **More** to display more call features.
2. Click **PIN List**.
3. On the **Add PIN List** page, configure the following settings:

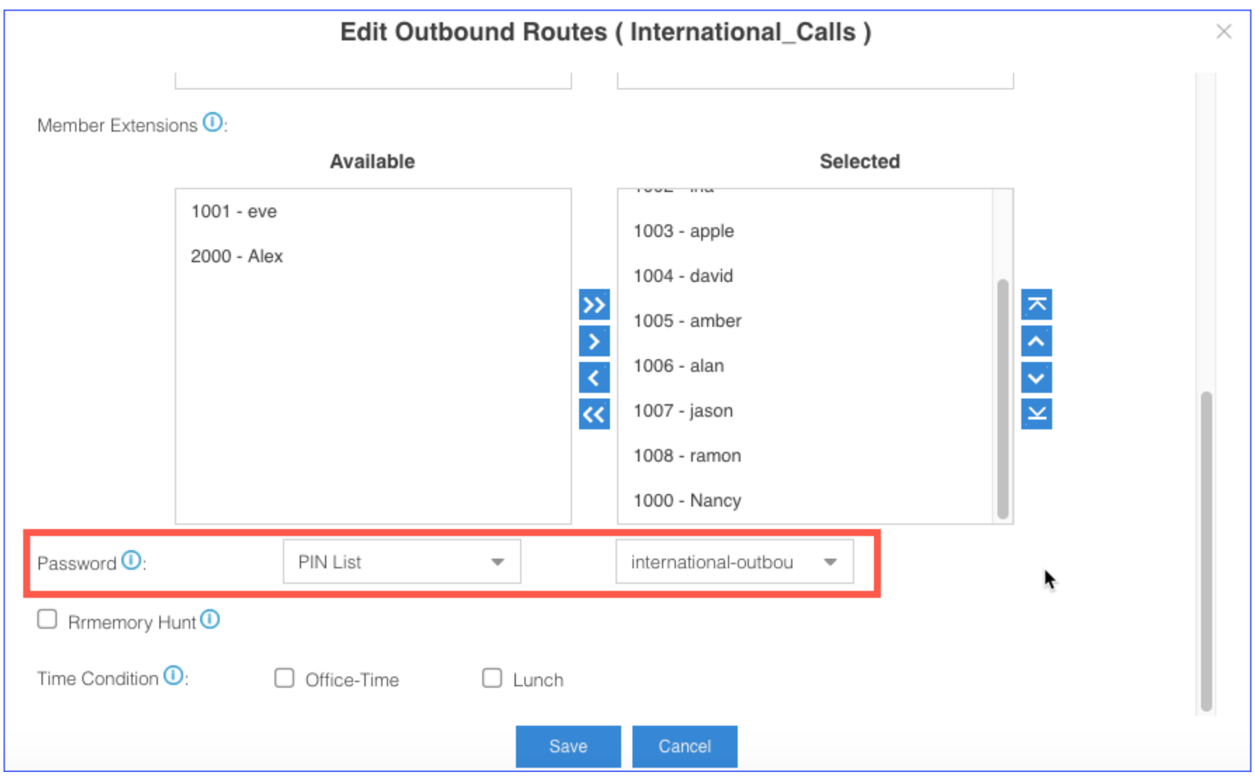


- **Name**: Set a name for the PIN list.
- **Record In CDR**: When a PIN code has been used, whether to display the PIN code in the relevant CDR.
- **PIN List**: Enter the PIN codes. Press **Enter** key to add multiple PIN codes.

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

Apply a PIN list

You can apply a PIN list to an outbound route or a DISA to restrict users dialling outbound calls. When a PIN list is applied to an outbound route or a DISA, users need to dial the correct PIN to place the outbound calls.



Blacklist/Whitelist

Yeastar Cloud PBX allows you to blacklist and whitelist IP addresses. This article briefly introduces the definitions and basic settings of blacklist and whitelist, and provides related configuration examples.

What is Blacklist and Whitelist

We briefly introduce the definitions of blacklist and whitelist as follows.

- Blacklist

  The blacklist is used to filter phone numbers. If a phone number is added to the blacklist, the system blocks incoming or outgoing calls for the phone number.
- Whitelist

  The whitelist is used to add trusted phone numbers. If a phone number is added to the whitelist, the system allows incoming or outgoing calls for the phone number.

  Note: The whitelist has a higher priority than the blacklist.

Blacklist/Whitelist Setting

Yeastar Cloud PBX supports system blacklist/whitelist and personal blacklist/whitelist. You can set a global system blacklist/whitelist to apply to all extensions. Extension users can also log in the PBX web interface by their accounts, and set blacklist/whitelist for their own extensions.

- **System Blacklist and Whitelist**

  Log in the PBX web interface as an administrator, and go to **Settings > PBX > Call Features > Blacklist/Whitelist** to set blacklist and whitelist.

  Yeastar Cloud PBX supports to block or allow three types of numbers:
  - **Inbound**: If blacklist type is set to **Inbound**, the number can not call in the system; if whitelist type is set to **Inbound**, the number can call in the system.
  - **Outbound**: Extension users can not call the number whose blacklist type is **Outbound**; extension users can call the number whose whitelist type is **Outbound**.
  - **Both**: Neither inbound calls nor outbound calls are allowed for the number whose blacklist type is **Both**; both inbound calls and outbound calls are allowed for the number whose whitelist type is **Both**.
- **Personal Blacklist and Whitelist**

Log in the PBX web interface by extension accounts, the extension users can view the system blacklist and whitelist that is set by the administrator.

Note: Extension users can add personal blacklist and whitelist for their extensions according to their needs.

- **Blacklist/Whitelist Priority**

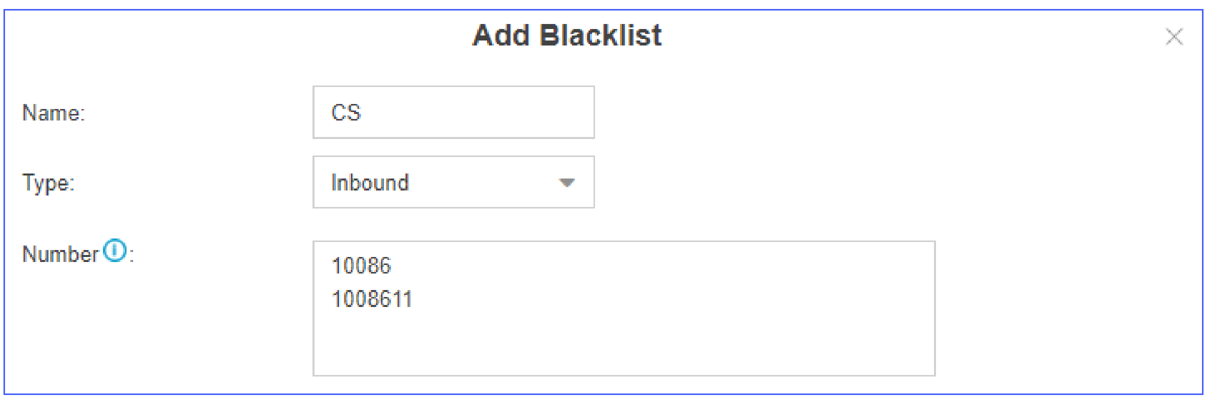
  Priority of blacklist/whitelist: system whitelist > system blacklist> personal whitelist > personal blacklist.

Blacklist Example

We demonstrate a few examples of blacklist as follows.

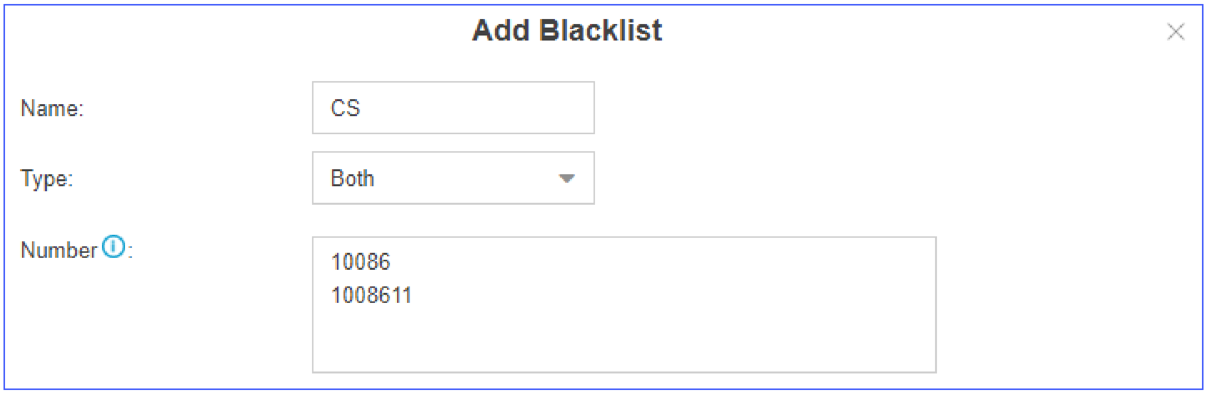
Prohibit inbound calls from external numbers

For example, 10086 and 1008611 are not allowed to call in PBX. You can add the two numbers to blacklist as follows.



Prohibit inbound calls and outbound calls

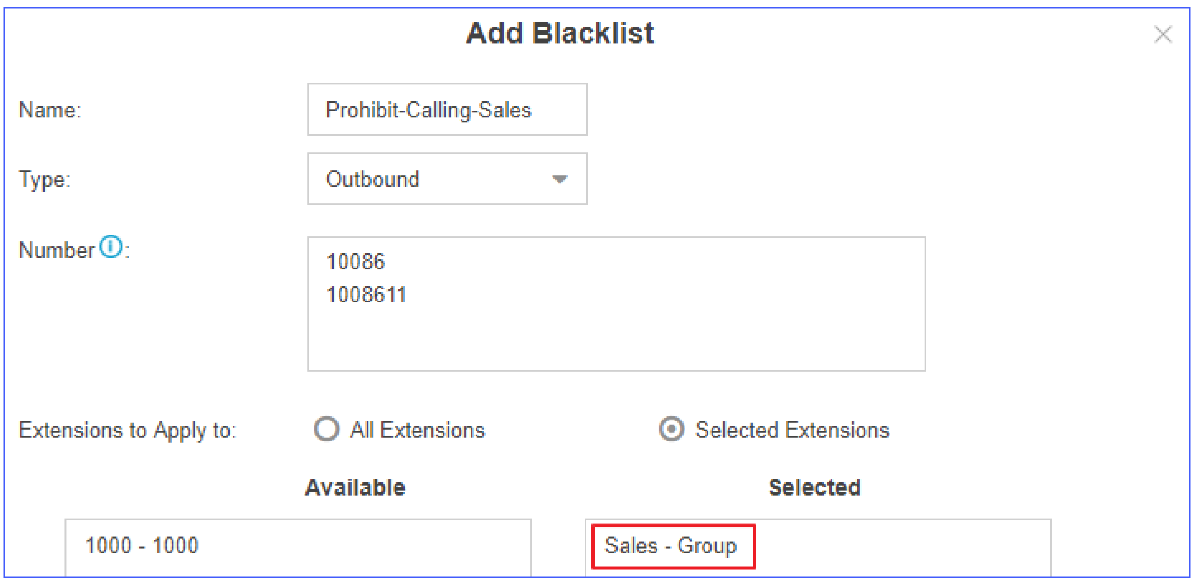
For example, 10086 and 1008611 are not allowed to call in PBX, and all extensions on PBX are not allowed to call out 10086 and 1008611.



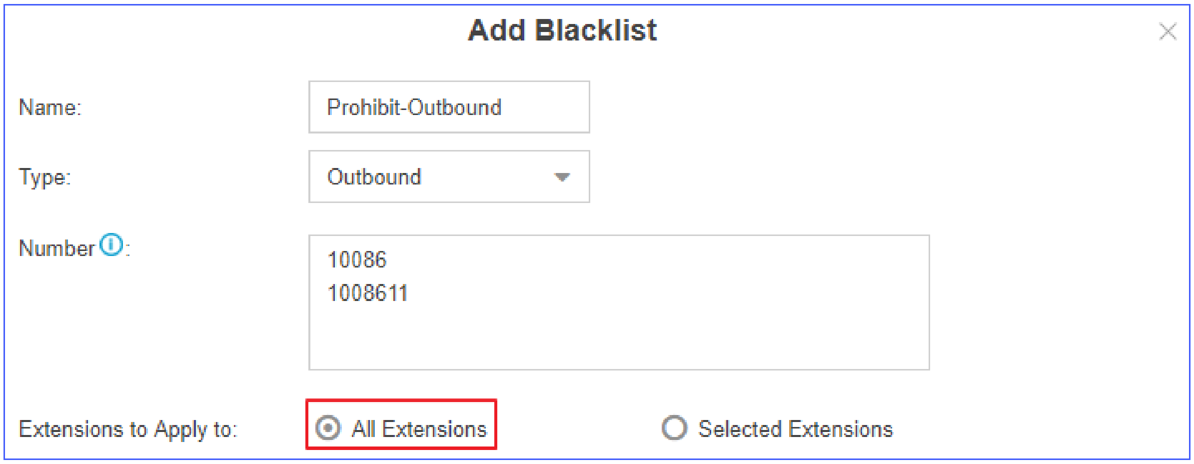
Prohibit selected extensions or extension groups from calling certain numbers

- Prohibit extension group (Sales) from calling 10086 and 1008611.

Note: You can add an extension group in advance for quick selection.

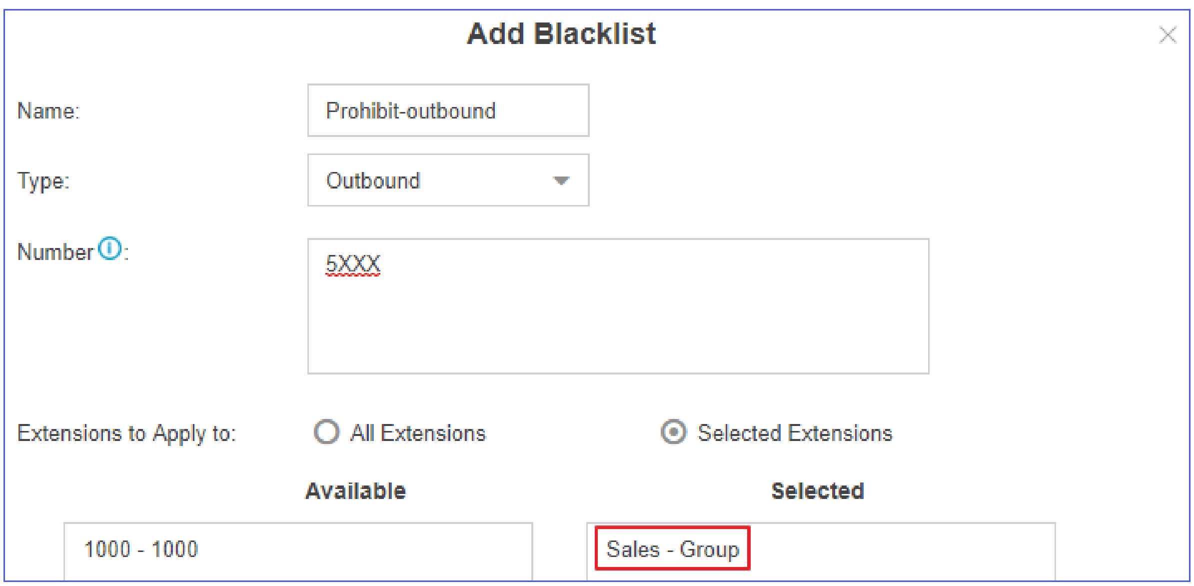


- Prohibit all extensions from calling 10086 and 1008611.



- **Prohibit extensions from calling numbers with specified extension format**

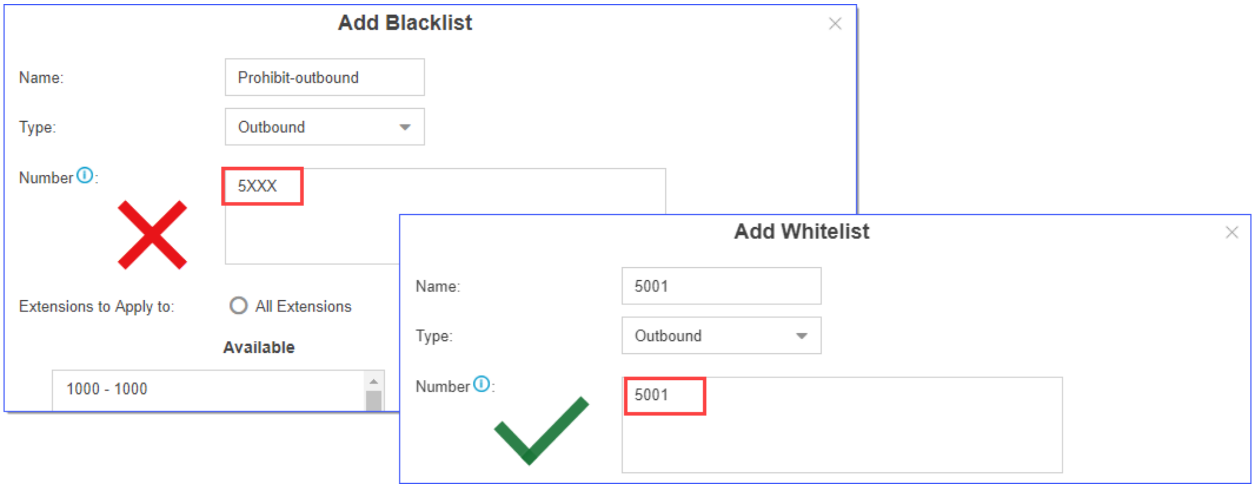
  For example, prohibit extension group (sales) from calling R&D team (all extension numbers are in the format 5XXX).



Whitelist Example

The whitelist has a higher priority than the blacklist, so you can use whitelist to filter trusted phone numbers from blacklist, and allow inbound/outbound calls for the phone numbers.

For example, assume you've added 5XXX (extension numbers of R&D team) to blacklist to prohibit sales from calling R&D teams, but you want to allow sales to call extension 5001. In this case, you can add 5001 to whitelist as follows.



Call Recording

Call Recording Overview

Yeastar Cloud PBX supports One Touch Recording and Auto Recording.

One Touch Recording

One Touch Reording, also known as On-demand Recording, allows users to dial *1 on their phones to record calls at any time.

For more detail of One Touch Recording, refer to One Touch Record.

Auto Recording

Auto Recording is a feature that enables the PBX to automatically record internal calls, external calls, and conference calls.

For more detail of Auto Recording, refer to Auto Recording.

One Touch Record

During a call, you can dial the One Touch Record feature code to start recording the call; dial the feature code again to stop the recording.

One Touch Record Feature Code

The default One Touch Record feature code is *1.

You can change the code via **Settings > PBX > General > Feature Code > One Touch Record**.

One Touch Record Prompt

By default, when a user dials *1 to record the call, the PBX will not play prompt to notify the other party that the call is being recorded.

To set One Touch Record prompt:

1. Go to **Settings > PBX > Voice Prompts > Prompt Preferences**.
2. In the **One Touch Record Start Prompt** field, select a custom prompt.

   When an extension user dial *1 to record the call, PBX will play the prompt to the other party.
3. In the **One Touch Recording End Prompt** field, select a custom prompt.

   When an extension user dial *1 to stop recording the call, PBX will play the prompt to the other party.
4. Click **Save** and **Apply**.

Auto Recording

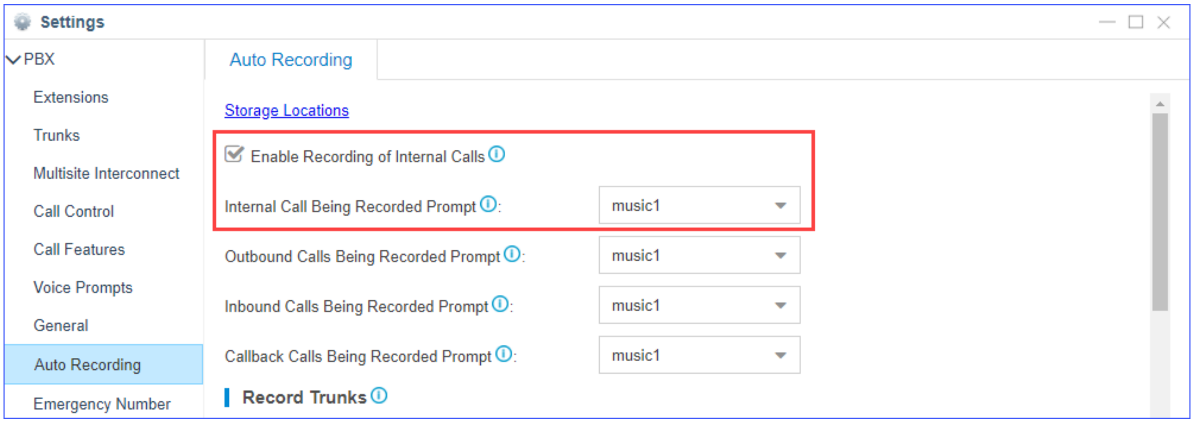
Auto Recording is a feature that enables the PBX to automatically record internal calls, external calls, and conference calls.

Set up Auto Recording

To use Auto Recording on your PBX, contact your PBX provider to buy the Recording capacity. By default, you can enjoy 500-minute recording time for free.

Set up Call Recording for Internal Calls

1. Go to **Settings > PBX > Recording**, check the option **Enable Recording of Internal Calls**.
2. Set the recording announcement for internal calls.

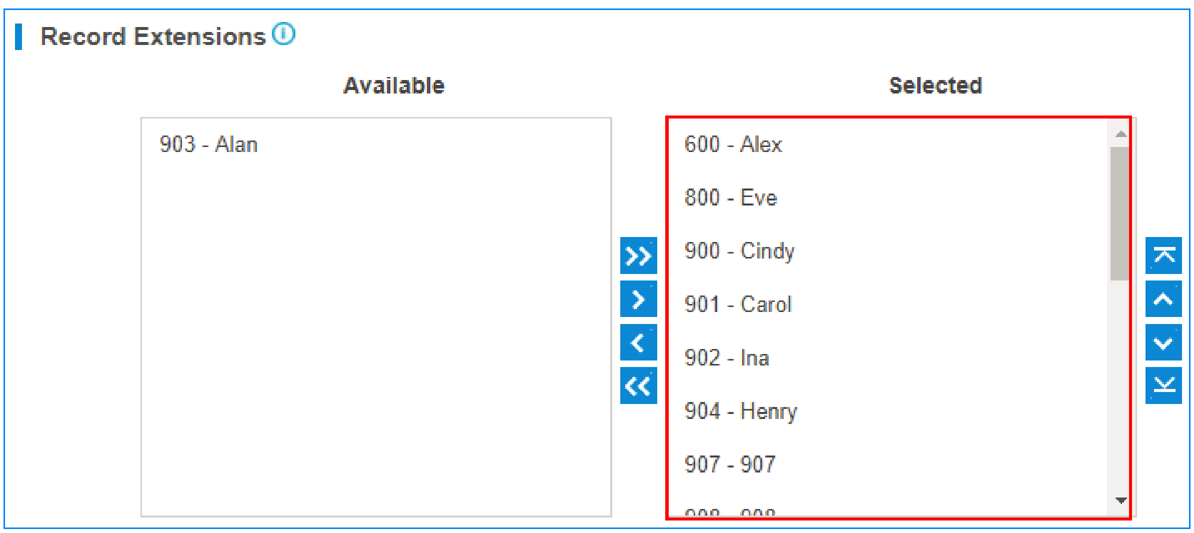


a. Upload a custom prompt to the PBX or record a custom prompt on the PBX.
b. Set **Internal Call Being Recorded Prompt** to your custom prompt.

The PBX will notify the called party that the call is being recorded.

3. In the **Record Extensions** section, select extensions to the **Selected** box.

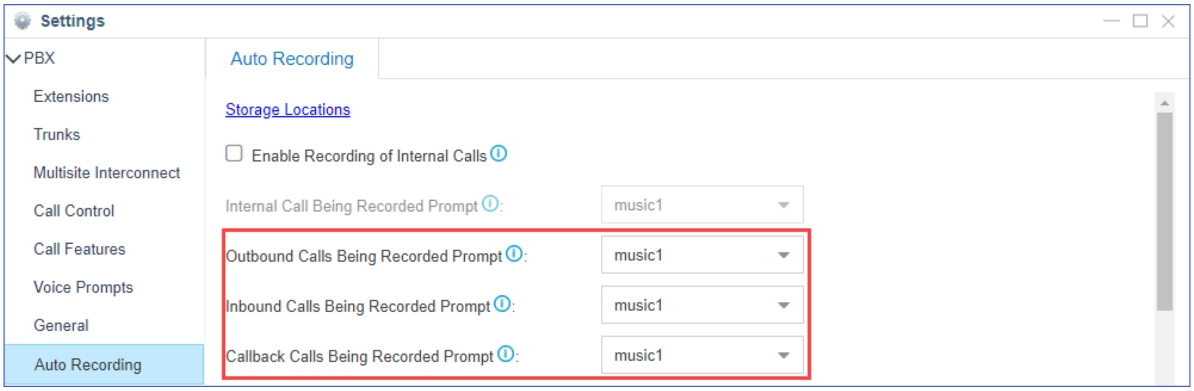
The selected extensions will be recorded.



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

Set up Call Recording for External Calls

1. Go to **Settings > PBX > Recording**, set the recording announcement for external calls.



a. Upload a custom prompt to the PBX or record a custom prompt on the PBX.
b. Set custom prompt for outbound calls, inbound calls, and callback calls.
- **Outbound Calls Being Recorded Prompt**: If the external call (outbound) has enabled call recording, this prompt will notify the external party that the call is being recorded.
- **Inbound Calls Recorded Prompt**: If the external call (inbound) has enabled call recording, this prompt will notify the external party that the call is being recorded.
- **Callback Calls Being Recorded Prompt**: If the external call (callback) has enabled call recording, this prompt will notify the external party that the call is being recorded.

2. In the **Record Trunks** section, select trunks to the **Selected** box.

The calls through the selected trunks will be recorded.

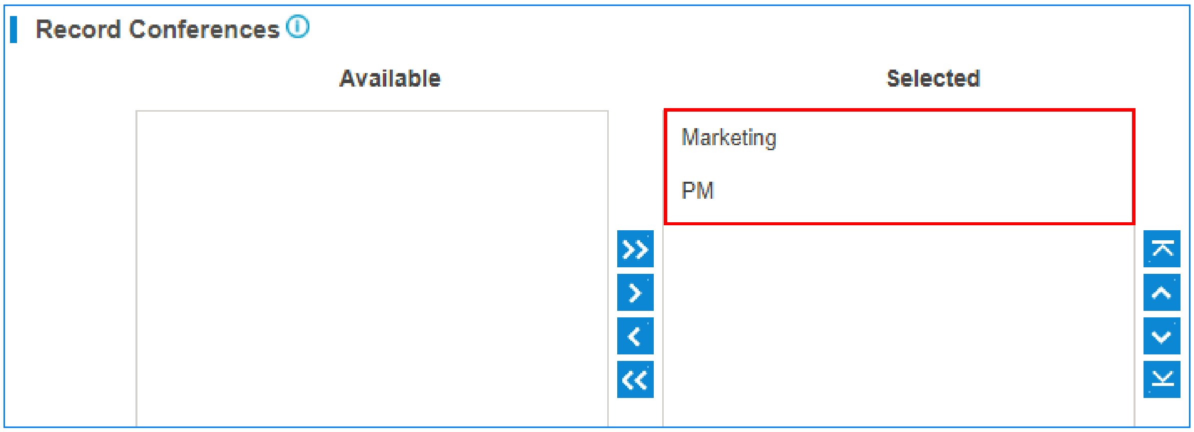
Note: If you have selected extensions in the **Record Extensions** section, the extensions' calls will be recorded no matter which trunks are used.

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

Set up Call Recording for Conference Calls

1. Go to **Settings > PBX > Recording**.
2. In the **Record Conferences** section, select conferences to the **Selected** box.

   The selected conferences will be recorded.



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

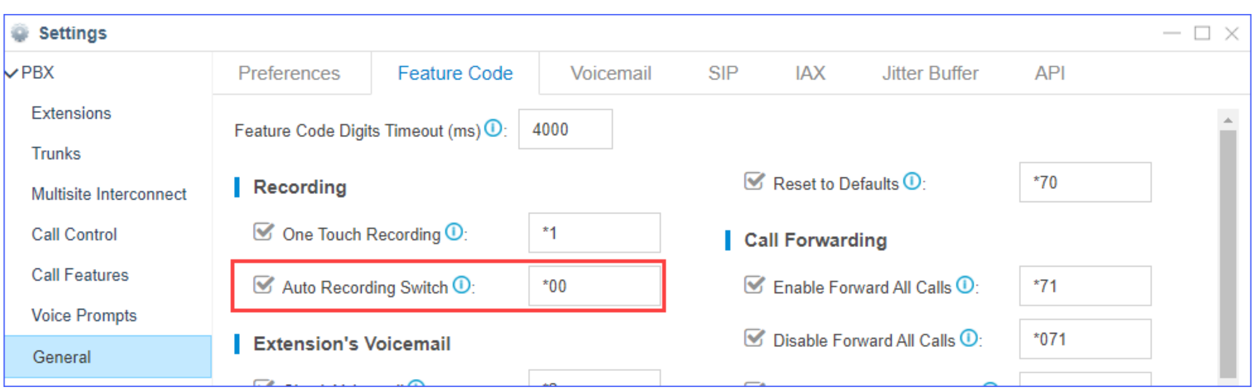
Pause/Resume Auto Recording

During an external call, the extension user can pause the Auto Recording and then resume the Auto Recording to avoid the sensitive personal information such as credit card details being recorded.

When you play the recording files, the paused part will be absent.

The default feature code to pause and resume Auto Recording is `*00`.

You can change the code in **Settings > PBX > General > Feature Code > Auto Recording Switch**.



During an external call, the extension user can dial feature code to pause and resume call recording.

1. Dial `*00` to pause the call recording.
2. Dial `*00` again to resume the call recording.

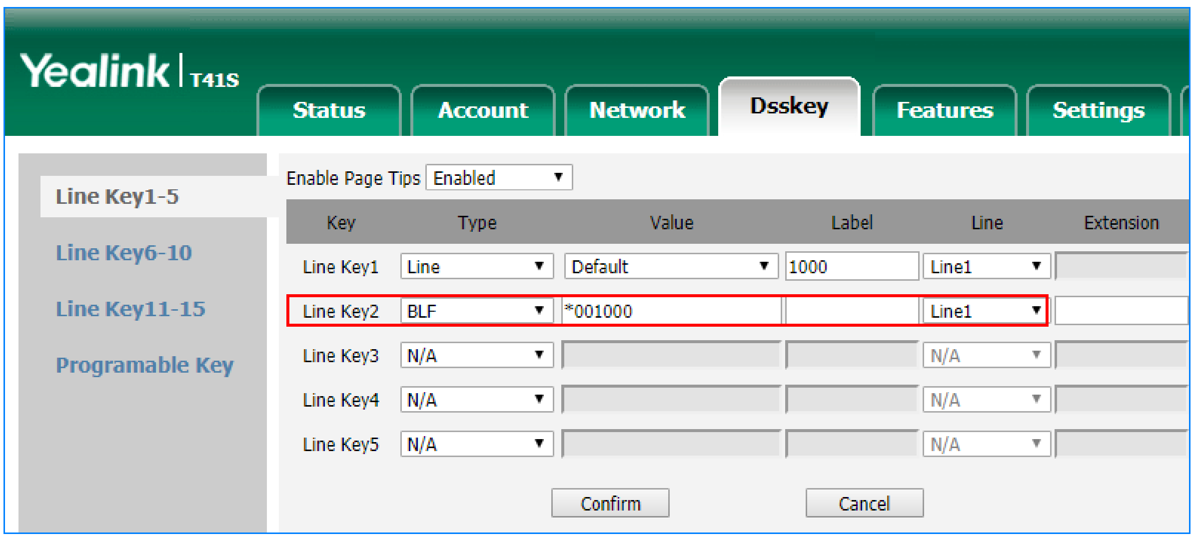
Related tasks
Monitor Auto Recording Status

Monitor Auto Recording Status

When you pause and resume the Auto Recording during a call, you may need to know if the call recording state is switched successfully or not. You can set a BLF key on your IP phone to monitor the auto recording status of your current call.

This topic is based on the Yealink T41S version 66.84.0.10.

1. Log in the phone web interface, go to **Dsskey > Line Key**.
2. Set a BLF key to monitor your own extension.
   In this example, your extension number is 1000, and the extension 1000 is registered on the phone Line 1.



   - **Type**: Set to **BLF**.
   - **Value**: The BLF key format is `*00{extension_number}`.

     In this example, set to **001000*.
   - **Label**: Optional. The label will be displayed on the phone screen.
   - **Line**: Choose the Line where your extension number is registered.
3. Click **Confirm**.

When the monitored extension is being recorded, the BLF LED will turn red.

When the monitored extension is not in a call or the Call Recording is paused, the BLF LED will turn green.

Related tasks
Pause/Resume Auto Recording

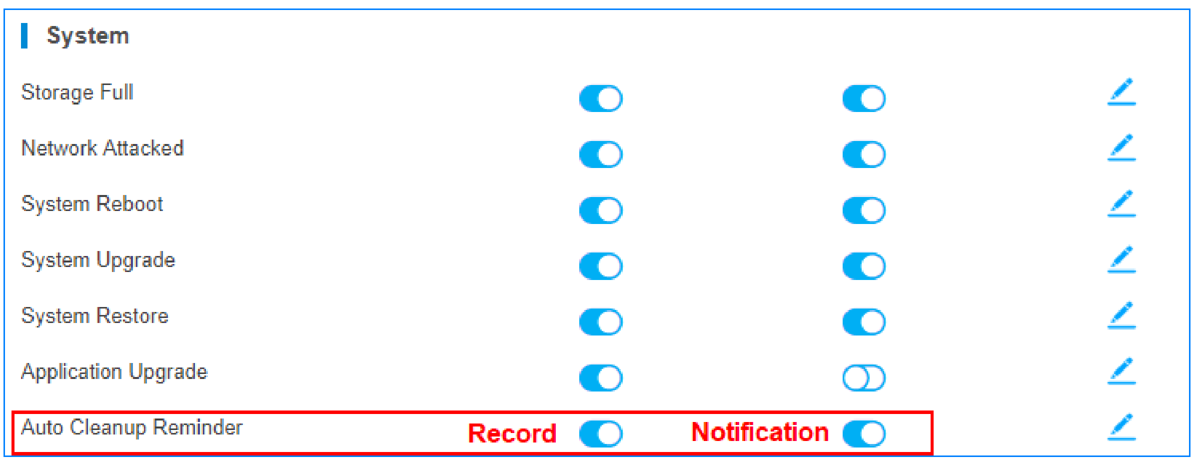
Auto Clean up Recording Files

When the recording capacity limit is reached, the PBX will automatically delete the oldest recording files. When 80% of the maximum recording capacity is reached, the PBX will send an email notification to you.

Enable 'Auto Cleanup Reminder'

To get informed of the recording usage, you can enable **Auto Cleanup Reminder**.

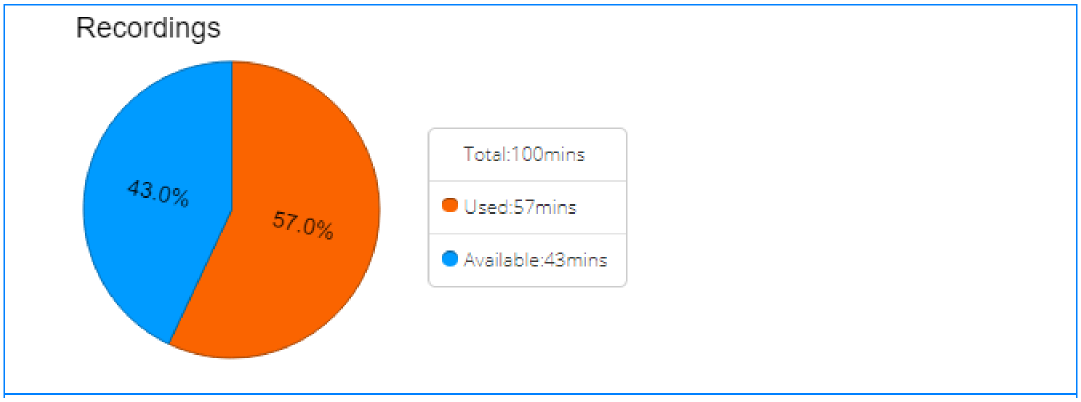
1. Go to **Settings > Event Center > Event Settings > System**, enable Notification and Record for **Auto Cleanup Reminder**.



    - **Record**: The event of Recording Auto Cleanup will be recorded in Event Log.
    - **Notification**: When the recording capacity is about to be reached, the PBX will send notification email to the Notification Contacts.
2. To modify the email template, click .

Check Recording Usage

Go to **Resource Monitor > Recording Usage** to check your Recording usage.

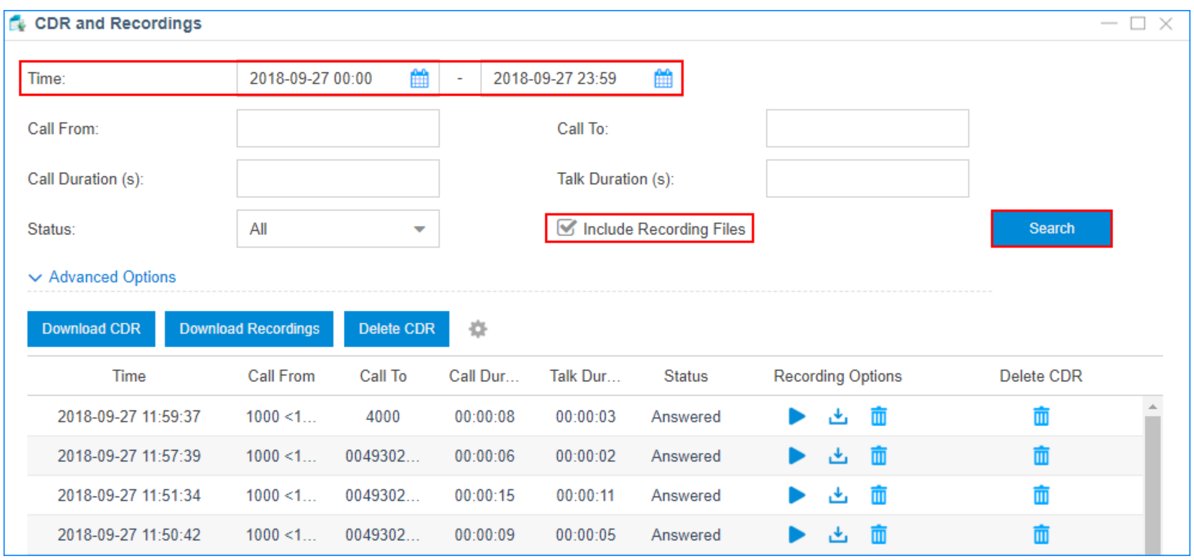


Manage Call Recording Files

Go to **CDR and Recordings** to search, play, download, or delete the recording files.

Search Recording Files

1. Set the search criteria **Time**.
2. Enable **Include Recording Files** to filter the records that have associated recording files.
3. Optional: Set other search criteria.
4. Click **Search**.



Download a Recording File

Click behind a recording log to download the recording file.

Play a Recording File

Click to play the recording file on web or play to an extension.

Delete a Recording File

Click behind a recording log to delete the recording file.

Grant Recording Permissions to Users

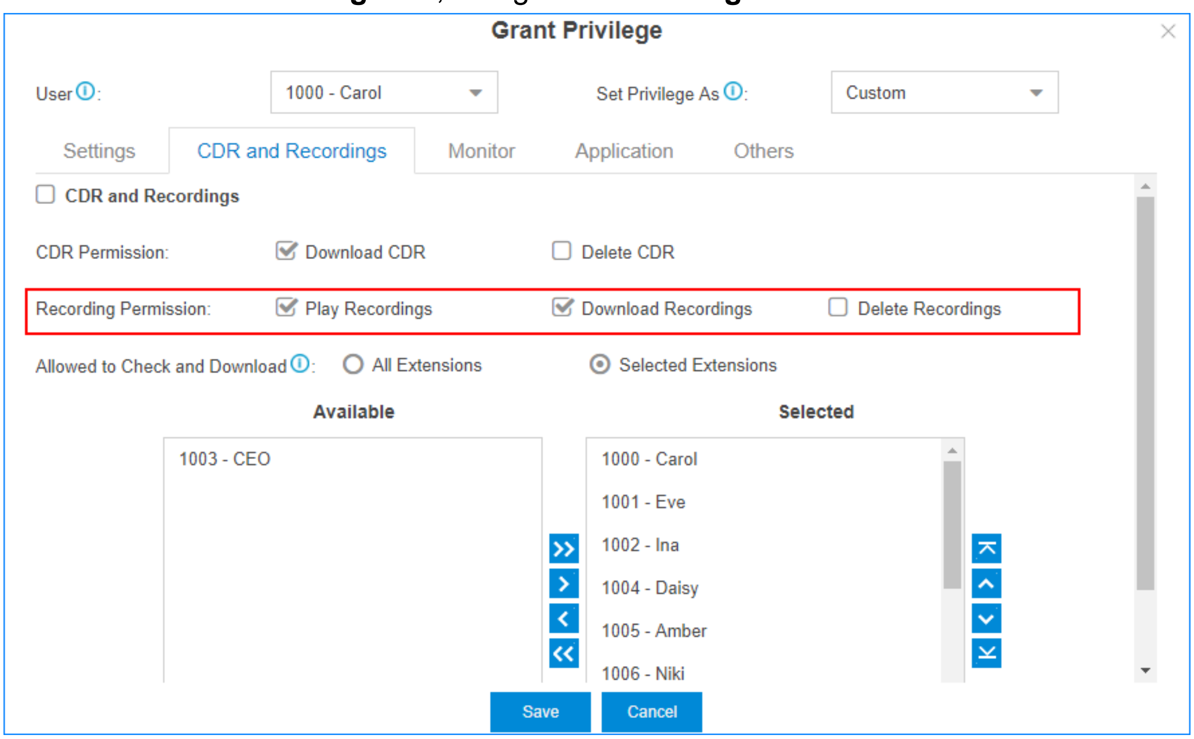
By default, only the super administrator has permission to manage the call recording files. The super administrator can grant recording permission to extension users and allow the users to play, download, and delete recording files.

1. Go to **Settings > System > User Permission**, click **Add**.

2. In the **User** drop-down list, select a user whom you want to grand permissions to.

3. Set **Set Privilege As**.
    - **Custom**: All permissions are disabled by default.
    - **Administrator**: All permissions are enabled by default.

4. Click **CDR and Recordings** tab, and grant **Recording Permission** to the user.



5. Set which extensions' recording files are allowed to play, download or delete.

- **All Extension**: The user can manage all the extensions' recording files.
- **Selected Extensions**: The user can manage only the selected extensions' recording files.

6. Click **Save**.
   When the user log in the PBX User Portal, he/she will have permission to manage recording files.

Voice Prompts

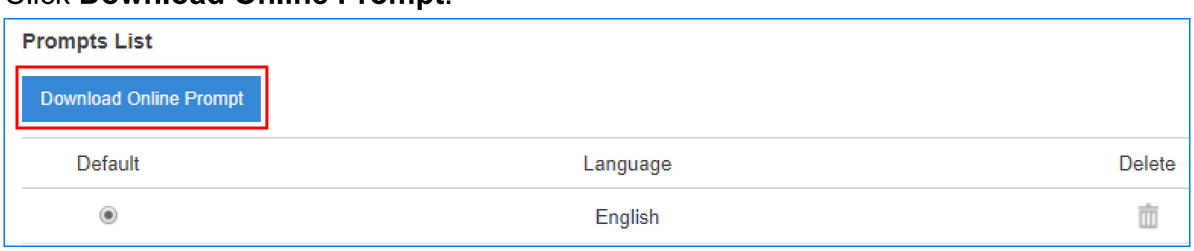
System Prompt

The default system prompt language is English. You can change the global system prompt, and if an extension user works in a foreign language, you can set a different system prompt for the user.

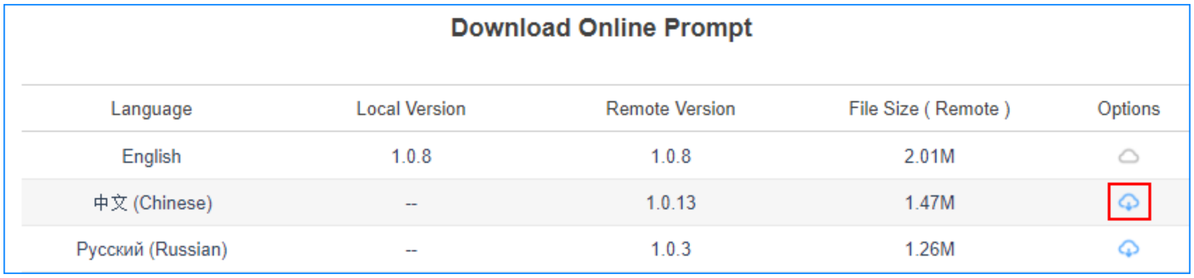
Change System Prompt

Yeastar have stored all the supported system prompts online. You can check the supported system prompts on the PBX web page, and download an online system prompt file, then change to the desired system prompt.

1. Go to **Settings > PBX > Voice Prompts > System Prompt**.
2. Click **Download Online Prompt**.



3. On the **Download Online Prompt** page, select your desired system prompt, click to download the file.
   After the file is downloaded, you can see the system prompt in **Prompt List**.



4. Set the downloaded system prompt as the default system prompt.

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

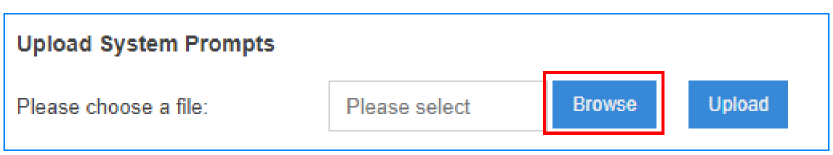
Customize System Prompt

You can upload your own system prompts to the PBX, so that users can hear the customized system prompts.

Contact Yeastar support to record your own system prompts.

1. Go to **Settings > PBX > Voice Prompts > System Prompt**.
2. In the **Upload System Prompts** section, click **Browse** to choose the system prompt file.

   Note: Upload the `.tar` file that is provided by Yeastar, or the system prompt won't work.



3. Click **Upload**.
   If the file is uploaded successfully, you can see the prompt file in the **Prompt List**.
4. Set the uploaded system prompt as the default system prompt.

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

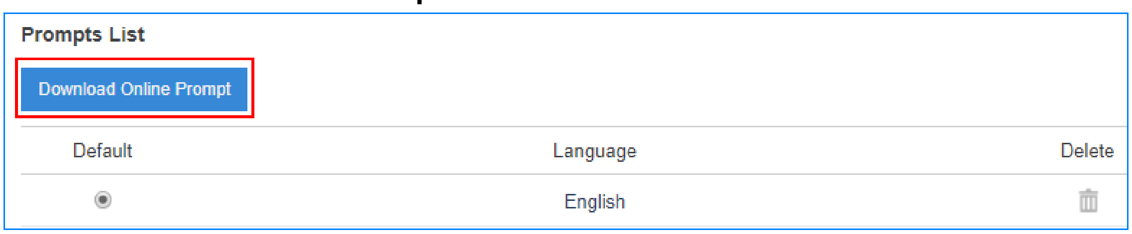
Change an Extension's System Prompt

If a user works in a foreign language, you can set a different system prompt for the extension user.

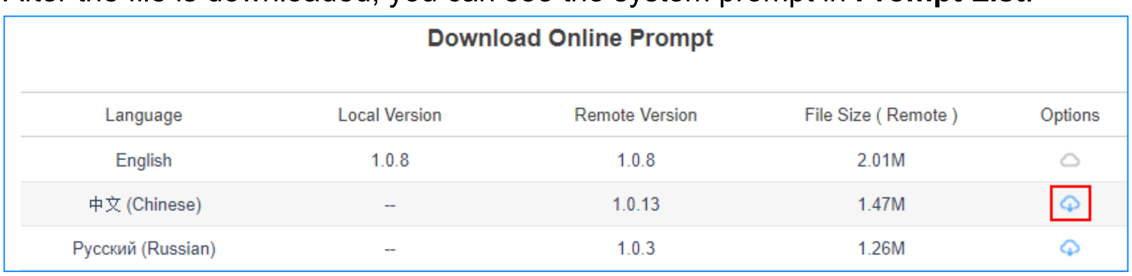
1. Download a system prompt for the extension user.

    a. Go to **Settings > PBX > Voice Prompts > System Prompt**.

    b. Click **Download Online Prompt**.

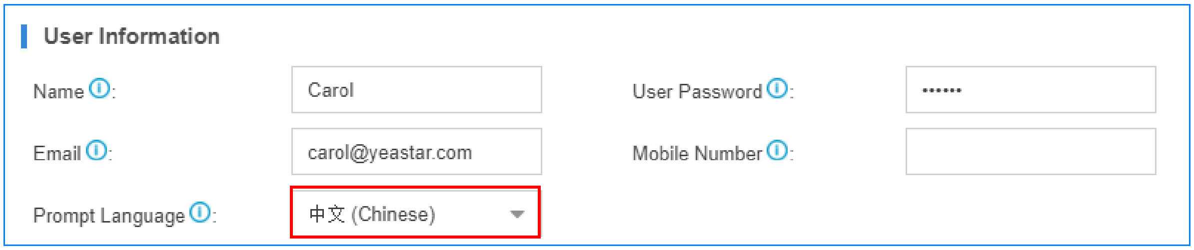


    c. On the **Download Online Prompt** page, select your desired system prompt, click to download the file.
    After the file is downloaded, you can see the system prompt in **Prompt List**.



2. Go to **Settings > PBX > Extensions**, select the desire extension, click .

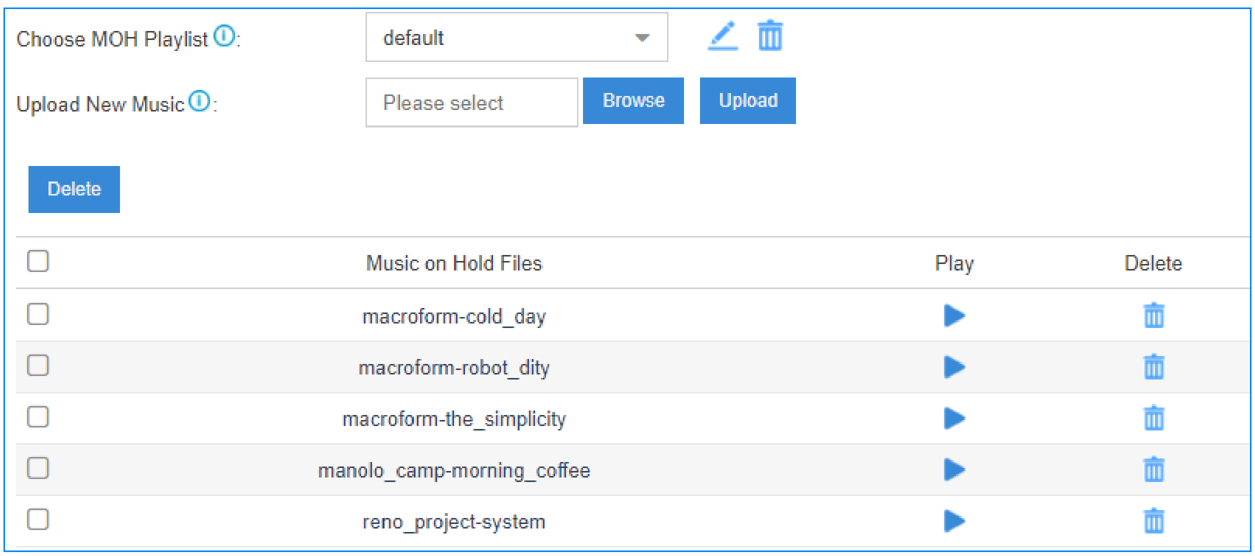
3. On the **Basic** page, set the **Prompt Language**.



Music on Hold (MoH)

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.

The PBX has a default MoH playlist, you can add MoH playlists and upload music files to the PBX.

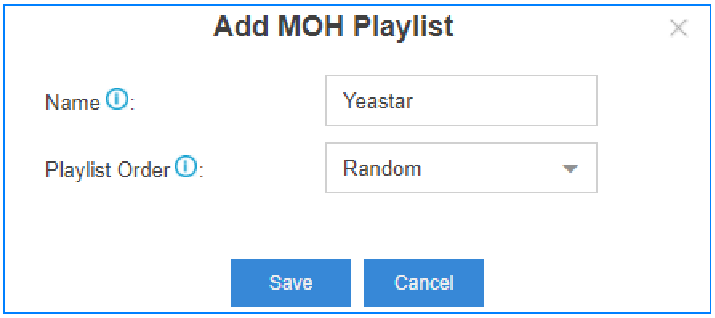


Note: The default MoH files are distributed under the Creative Commons Attribution-ShareAlike3.0 license through explicit permission from their authors.

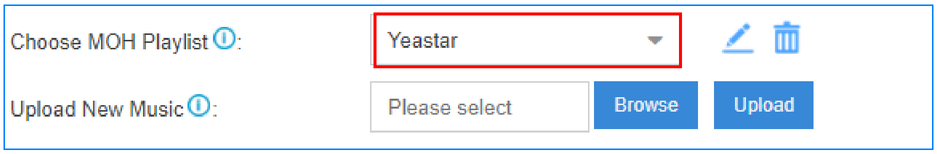
Add a Custom MoH Playlist

You can add a custom MoH playlist and upload your audio files to the PBX.

1. Go to **Settings > PBX > Voice Prompts > Music on Hold**, click **Create New Playlist**.
2. On the configuration page, set the playlist name and the playlist order, click **Save**.



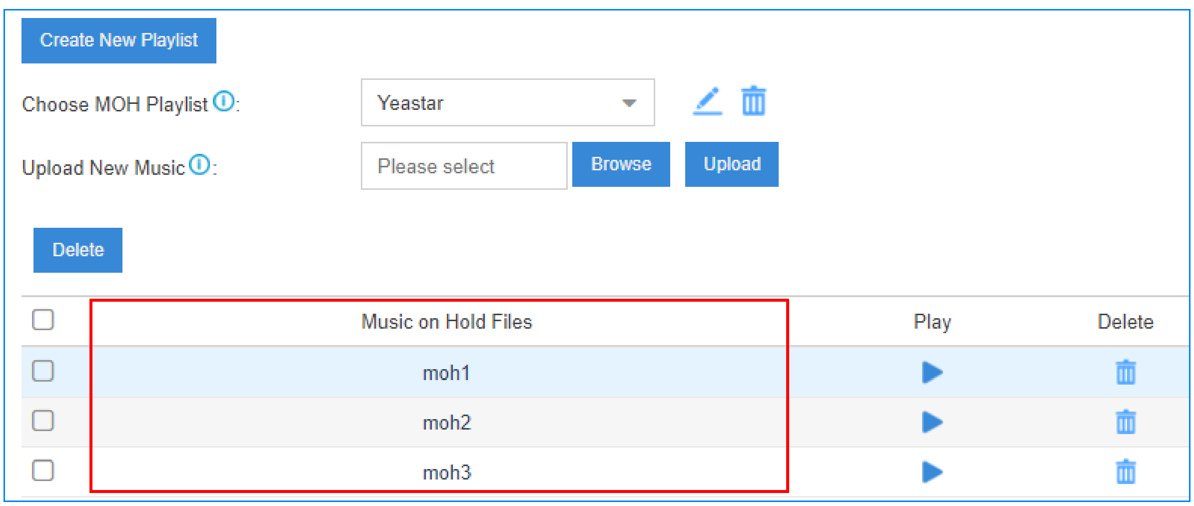
3. On the **Music On Hold** page, choose the new created playlist.



4. Click **Browse** to choose an audio file from your local PC, then click **Upload**.

   Note: The uploaded file should meet the audio file requirements.

5. Repeat the step 4 to add another audio file.
   You can see the uploaded audio files in the MoH list.



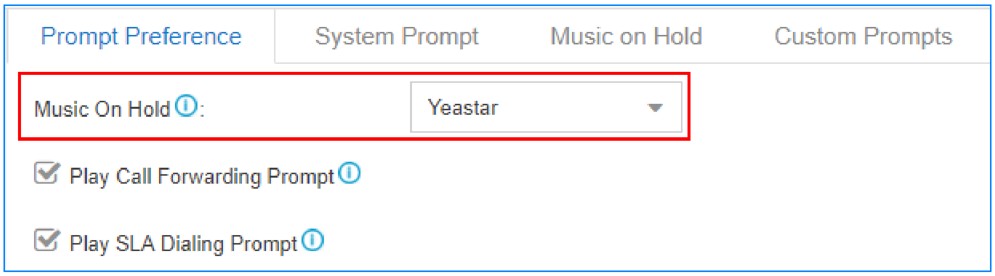
Related concepts
Requirements of Custom Audio Files

Related tasks
Change the MoH Playlist
Convert Audio Files via WavePad
Convert Audio Files Online

Change the MoH Playlist

To change the MoH playlist, you need to first add a MoH playlist and upload your audio files to the PBX.

1. Go to **Settings > PBX > Voice Prompts > Prompt Preference**.
2. Select a MoH playlist from the drop-down menu of **Music On Hold**.



The PBX will play the select MoH playlist when a user is held in a call.

Related tasks
[Add a Custom MoH Playlist]

Custom Prompt

The default voice prompts and announcements in the system are suitable for almost every situation.

However, you may want to use your own voice prompt to make it more meaningful and suitable for your case. In this case, you need to upload a custom prompt to the system or record a new prompt and apply it to the place you want to change.

Requirements of Custom Audio Files

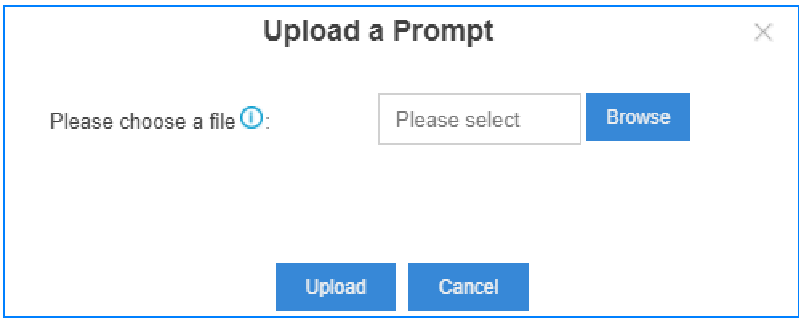
You can upload your audio file to the PBX, the audio file should meet the following requirements.

Option	Requirement
File Format	`WAV`, `wav`, or `gsm` file. • gsm 6.10 8kHz, Mono, 1Kb/s • alaw 8kHz, Mono, 1Kb/s • ulaw 8kHz, Mono, 1Kb/s • pcm 8kHz,Mono,16Kb/s
File Name	Should NOT contain special characters.

Option	Requirement
File Size	Smaller than 8MB.

Upload a Custom Prompt

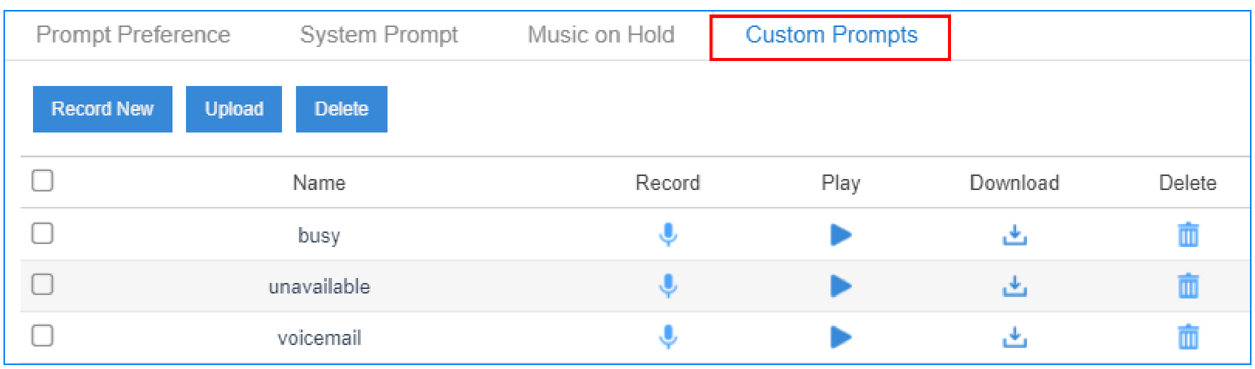
1. Go to **Settings > PBX > Voice Prompts > Custom Prompts**, click **Upload**.
2. On the configuration page, click **Browse** to choose your audio file.



Note: The uploaded file should meet the audio file requirements.

3. Click **Upload** to start uploading the file.

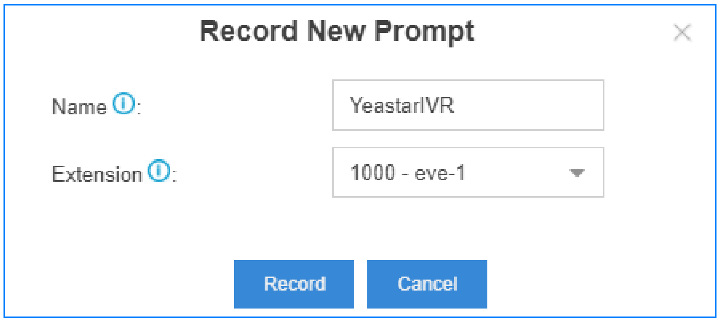
After the file is uploaded, you can see the file on the **Custom Prompts** page.



Record a Custom Prompt

You can use an extension to record custom prompts.

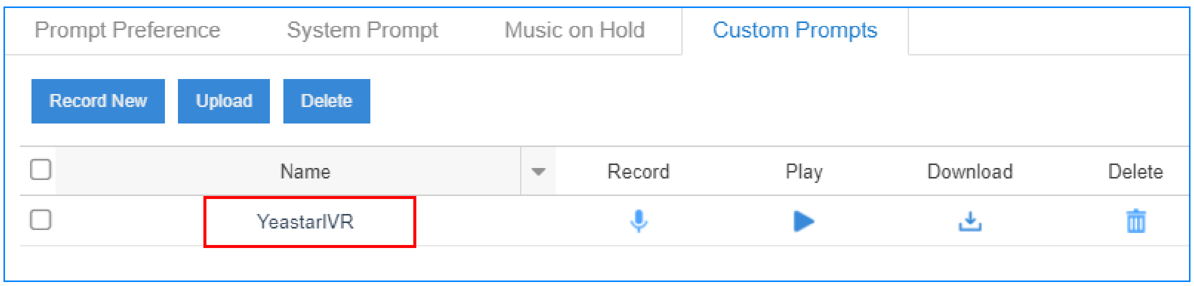
1. Go to **Settings > PBX > Voice Prompts > Custom Prompts**, click **Record New**.
2. On the configuration page, set the prompt name and select an extension to record the prompt.



3. Click **Save**.
   The selected extension will ring.

4. Record your prompt on the phone. When done, press the # key or hang up.

5. Refresh the **Custom Prompts** page, you can see the saved prompt file.



You can click to play the prompt, and decide whether to save it or not. If you are not satisfied with the prompt, click to record again.

Related tasks
Play a Custom Prompt

Play a Custom Prompt

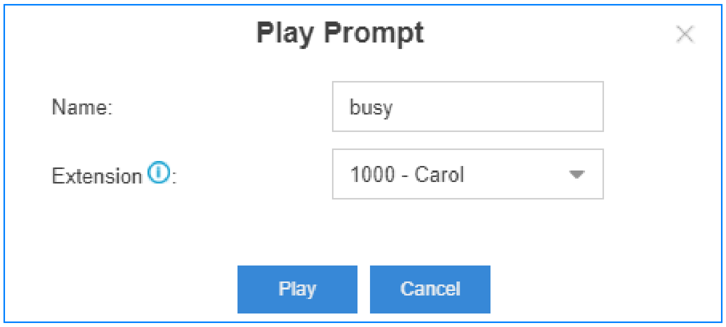
After you upload a custom prompt or record a custom prompt, you can select an extension to play the prompt.

Note: We recommend that you play your custom prompts before you apply the custom prompts to IVR, MoH, or other places.

1. Go to **Settings > PBX > Voice Prompts > Custom Prompts**.

2. In the Custom Prompts list, choose a prompt, click .

3. On the configuration page, choose an extension to play the prompt.



4. Click **Play**.
   The selected extension will ring.

5. Pick up the phone to listen to the prompt.

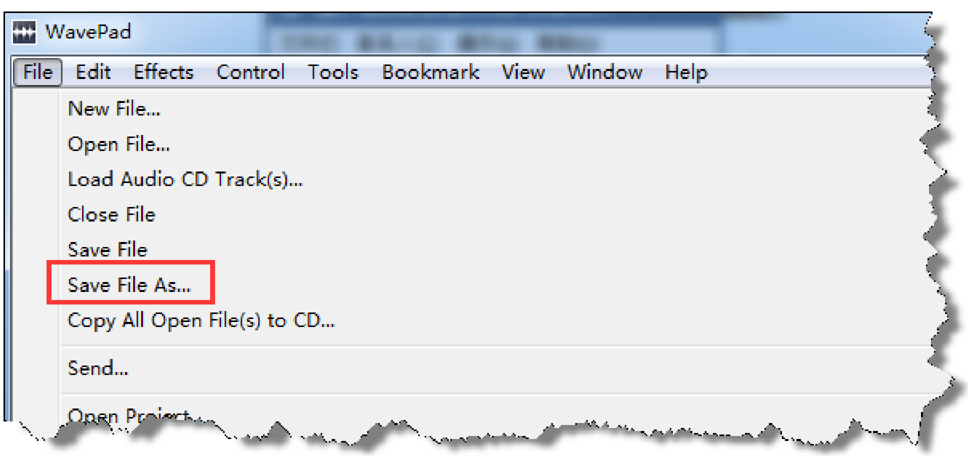
Related tasks
Upload a Custom Prompt
Record a Custom Prompt

Convert Audio Files via WavePad

WavePad is audio editing software, you can convert audio files via WavePad, then upload the audio files to your PBX.

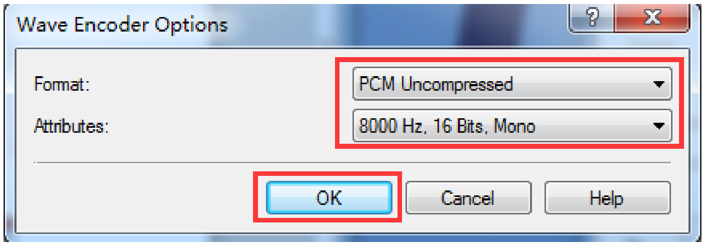
1. Launch WavePad, open your audio file.

2. Click **File > Save File As**.



3. Set the **Save as type** to `.wav` or `.gsm`, click **Save**.

4. For the `.wav` type, set the encoder options according to the requirements of custom audio files, click **OK**.

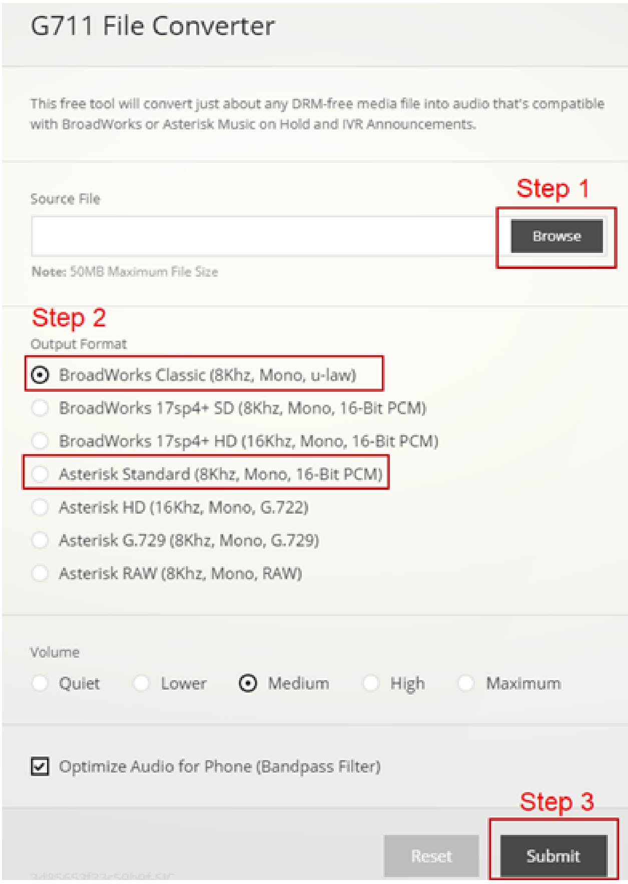


Related tasks
[Convert Audio Files Online](#)

Convert Audio Files Online

You can quickly convert your audio files via G711 File Converter online.

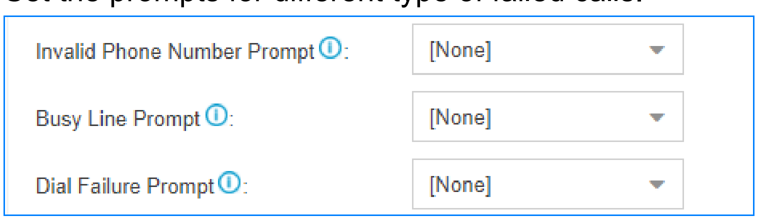
1. Visit [g711.org](#).
2. Click **Browse** to upload your audio file.
3. Set the **Output Format**.
   We recommend **BroadWorks Classic** or **Asterisk Standard**.
4. Click **Submit** to start converting the file.



Set Prompts for Failed Calls

A user may fail to make outbound calls due to many reasons, such as the trunk is busy, no trunk available, or invalid number. You can set different prompts to inform the user why the call fails.

1. Go to **Settings > PBX > Voice Prompts > Prompt Preference**.
2. Set the prompts for different type of failed calls.



- **Invalid Phone Number Prompt**: The PBX will play the prompt when the dialed number is invalid.
- **Busy Line Prompt**: The PBX will play the prompt when the trunk used is busy.
- **Dial Failure Prompt**: The PBX will play the prompt if no trunk is available to call out.

System Management

System General Settings

The system general settings can be applied globally to Yeastar Cloud PBX

System Preference

Configure the preferences settings that will be applied globally to the system.

Go to **Settings > PBX > General > Preferences** to configure the system preferences.

General Preference

Table 2. Descriptions of General Preference

Option	Description
Max Call Duration	Select the global maximum call duration. **Note:** The precedence of **Max Call Duration(s)** (Global v.s. Extension): • For internal calls: The **Max Call Duration(s)** setting of the caller's extension takes precedence. • For outbound calls: The **Max Call Duration(s)** setting of the caller's extension takes precedence. • For inbound calls: The global **Max Call Duration(s)** setting takes precedence.
Attended Transfer Caller ID	The Caller ID that will be displayed on the recipient's phone. For example, Phone A (transferee) calls Phone B (transfer), and Phone B transfers the call to Phone C (recipient). If set to Transfer, the Caller ID displayed will be Phone B's number; if set to Transferee, Phone A's number will be displayed.
Flash Event	Set which event will be triggered by pressing the hook flash: • 3-way Calling • Call Transfer
Virtual Ring Back Tone	Once enabled, when the caller calls out with cellular trunks, the caller will hear the virtual ring back tone generated by the system before the callee answers the call.

Option	Description
Distinctive Caller ID	When the incoming call is routed from Ring Group, Queue or IVR, the Caller ID would display where it comes from.
Match Route Permission When Seizing a Line	If checked, when users seize a line to place an outbound call, the call will succeed only when the route permission is matched.
FXO Mode	Select a mode to set the On Hook Speed, Ringer Impedance, Ringer Threshold, Current Limiting, TIP/RING voltage, adjustment, Minimum Operational Loop Current, and AC Impedance as predefined for your country's analog line characteristics. The default setting is FCC for USA.
Tone Region	Select your country or nearest neighboring country to enable the default dial tone, busy tone, and ring tone for your region.
DTMF Duration	Set the duration of a DTMF tone on the FXO trunk.
DTMF Gap	Set the interval between each DTMF tone on the FXO trunk.

Extension Preference

Below are default extension ranges. You can change the extension range according to your needs.

Note: PBX treats Ring Group, Paging Group, Conference, Queue as extensions. Extension users can dial the extension numbers to reach them directly.

Extension Type	Default Range
User Extensions	1000 - 5999
Account Trunk	6100 - 6199
Ring Group Extensions	6200 - 6299
Paging Group Extensions	6300 - 6399
Conference Extensions	6400 - 6499
IVR Extensions	6500 - 6599
Queue Extensions	6700 - 6799

Feature Code

Feature codes are used to enable and disable certain features available in the Yeastar Cloud PBX. Extension users can dial feature codes on their phones to use that particular feature.

Go to **Settings > PBX > General > Feature Code** to view or change the feature code settings.

- **Feature Code Digit Timeout**: The timeout to input next digit. The default is 4000 ms.

Default Feature Codes

Recording	
One Touch Record	*1
Call Recording Switch	*00
Call Forwarding	
Reset to Defaults	*70
Enable Forward All Calls	*71
Disable Forward All Calls	*071
Enable Forward When Busy	*72
Disable Forward When Busy	*072
Enable Forward No Answer	*73
Disable Forward No Answer	*073
Voicemail	
Check Voicemail	*2
Voicemail for Extension	**
Voicemail Main Menu	*02
Transfer	
Blind Transfer	*03
Attended Transfer	*3
DND	
Enable Do Not Disturb	*74
Disable Do Not Disturb	*074
Call Pickup	
Call Pickup	*4
Extension Pickup	*04
Time Condition	
Time Condition Override	*8
Intercom	

Recording	
Intercom	*5
Call Monitor	
Listen	*90
Whisper	*91
Barge-in	*92
Call Parking	
Call Parking	*6
Directed Call Parking	*06
Parking Extension Range	6900-6999

SIP Settings

The SIP configurations require professional knowledge of SIP protocol, incorrect configuration may cause calling issues on the SIP extensions and SIP trunks.

Go to **Settings > PBX > General > SIP** to configure the SIP settings.

SIP General Settings

Option	Description
UDP Port	UDP Port used for SIP registrations. The default is 5060.
TCP Port	TCP Port used for SIP registrations. The default is 5060.
Registration Timers	
Max Registration Time	Maximum duration (in seconds) of incoming registrations and subscriptions. The default is 3600 seconds.
Min Registration Time	Minimum duration (in seconds) of incoming registration and subscriptions. The default is 60 seconds.
Qualify Frequency	How often to send SIP OPTIONS packet to SIP device to check if the device is up. The default is 30 per second.
Outbound SIP Registrations	
Registration Attempts	The number of registration attempts before giving up (0 for no limit).
Default Incoming/Outgoing Registration Time	Default duration (in seconds) of incoming/outgoing registration. The default is 120 seconds. **Note:** The actual duration needs to minus 10 seconds from the value you filled in.

Option	Description
Subscription Timer	
Max Subscription Time	Maximum duration (in seconds) of incoming subscriptions. The default is 3600 seconds.
Min Subscription Time	Minimum duration (in seconds) of incoming subscriptions. The default is 90 seconds.

SIP Codec

A codec is a compression or decompression algorithm that used in the transmission of voice packets over a network or the Internet.

Codec Selection

Yeastar Cloud PBX supports G711 a-law, u-law, GSM, H261, H263, H263P, H264, SPEEX, G722, G726, ADPCM, G729A, MPEG4, and iLBC.

Note:

- You need to choose at least one same code on the PBX and on your phones, or there may be a problem of the call.
- If you want to make video calls, you need to select H261, H263, H263P, H264 or MPEG4 codec on the PBX and on your phones.

iLBC Settings

The iLBC codec supports two modes: 20ms and 30ms frame length modes,

To get better voice quality, you need to set the iLBC mode according to your SIP endpoints.

Note: Linkus uses iLBC 20ms mode. When Linkus is enabled, this option is switched to 20ms mode automatically.

TLS Settings

Option	Description
Enable TLS	Check the checkbox to enable TLS.
TLS Port	TLS Port used for SIP registrations. The default is 5061.
TLS Client Method	Specify protocol for outbound client connections. The default is sslv2.

Session Timer

A periodic refreshing of a SIP session that allows both the user agent and proxy to determine if the SIP session is still active.

Option	Description
Session-timers	Choose the session timers mode on the system: • **No**: Do not include “timer” value in any field • **Supported**: Include “timer” value in Supported header • **Require**: Include “timer” value in Require header • **Forced**: Iclude “timer” value in both pportednd equired header. The default is **Supported**.
Session-Expires	The max refresh interval in seconds.
Min-SE	The min refresh interval in seconds, it must not be less than 90.

Qos

QoS (Quality of Service) 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 interference from other lower priority traffic.

When the network capacity is insufficient, QoS could provide priority to users by setting the value.

Option	Description
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 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 don't work.

Option	Description
No T.38 Attributes in Re-invite SDP	If this option is selected, SDP re-invite packet will not contain T.38 attributes.

Option	Description
Error Correction	Enable or disable Error Correction for the fax.
T.38 Max BitRate	Adjust the max BitRate for T.38 fax.

Advanced SIP Settings

Option	Description
User Agent	Change the User-Agent field.
Send Remote Party ID	Whether to send Remote-Party-ID in SIP header or not. **Note:** This configuration only take effects on internal calls. To set up for external calls, configure the **Advance** settings of SIP trunk.
Send P Asserted Identify	Whether to send P-Asserted-Identify in SIP header or not. **Note:** This configuration only take effects on internal calls. To set up for external calls, configure the **Advance** settings of SIP trunk.
Send Diversion ID	Whether to send Diversion in SIP header or not. If this option is selected, the Diversion value will be extension number. **Note:** This configuration only take effects on internal calls. To set up for external calls, configure the **Advance** settings of SIP trunk.
Support Early Media	Whether to support Early Media or not.
All Busy Mode for SIP Forking	• Check this option: When one of the terminals that register the same extension number is busy in a call, the other terminals will not receive calls. • Uncheck this option: When one terminal is busy, the other terminals will still be able to make and receive calls.

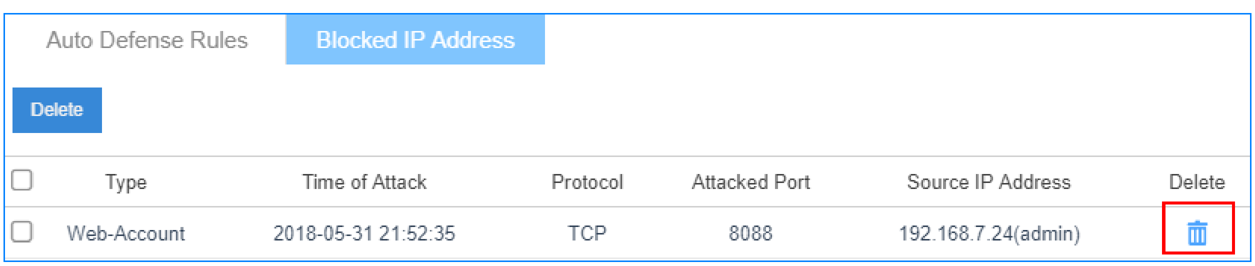
Option	Description
Inband Progress	This Inband Progress setting applies to all the extensions. **Note:** To configure global Inband Progress setting, you need to contact Yeastar support to configure a custom config file. • Check this option: PBX will send a 183 Session Progress to the extension when 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.
Get Caller ID From	Decide the system will retrieve Caller ID from which header field.
Get DID From	Decide the system will retrieve DID from which header field. **Note:** If Remote-Party-ID is selected but the SIP trunk doesn't support this, the system will retrieve DID fron INVITE header.
100rel	Whether to support 100rel or not.
Support Message Request	Whether to support SIP Message Request or not.
Maxptime	Select or enter the Maxptime value.

Security

Blocked IP Address

The PBX will block an IP address for too many failed login attempts, too many failed registration attempts, or too many failed authentications for Auto Provisioning.

The blocked IP addresses would be listed in the Blocked IP Address table. If a trusted IP address was blocked by the PBX, you can go to **Settings > System > Security > IP Auto Defense > Blocked IP Address** to delete the IP address.



Service

All the PBX service statuses and ports are displayed on the security Service page.

Go to **Settings > System > Security > Service** to configure the service settings.

Option	Description
Auto Logout Time (min)	After the set time of inactivity, the session will automatically log out. The default time is 15 minutes.
SIP UDP Port	SIP registration port. The default SIP UDP port is 5060.
Enable SIP TCP	Whether to enable SIP TCP or not. The default port is 5060.
Enable SIP TLS	Whether to enable SIP TLS or not. The default port is 5061.

User Permission

By default, the extension users can log in the system and check their own settings and CDR. You can set different permission to the users according to their roles and duty.

User Types on the PBX

Super Admin

Super Admin has the highest privilege. The super administrator can access all pages on S-Series Web and make all the configurations on the system.

- Username: `admin`

Administrator or Custom User

Administrator or Custom User is created by the Super Admin. The Super Admin sets the privileges for those users according to their roles and duty.

- Username: The extension number or the email address of the extension user.

Note:

- **Administrator** and **Custom User** can have the same permission. The different between the two role type:
    - **Administrator**: All permissions are enabled by default.
    - **Custom User**: No permission is enabled by default.
- **Administrator** and **Custom User** do not have permission to configure **User Permission**.

Configure User Permission

To grand more privilege for a user or change the user's privilege, you need to configure the User Permission on PBX.

Scenarios

In the following scenarios, you may need to add permissions for the extension users according to their roles.

- For an HR, he/she may need the permission to add extension, configure extension's outbound route privilege when there are new staffs.
- For a supervisor, he/she will have permission to check the CDR and recordings, and have no permission to configure the system or other extensions.

Procedures

1. Log in the PBX web interface by the super admin account, go to **Settings > System > User Permission**, click **Add**.
2. On the configuration page, select the **User**.
3. Set the **Set Privilege As**.
    - **Administrator**: All the permissions are enabled for the user by default.
    - **Custom User**: No permission is enabled for the user by default.
4. Click the **Settings**, **CDR and Recordings**, **Monitor**, **Application**, **Contacts**, and **Others** tabs, and check or uncheck the relevant options for the user.
5. Click **Save** and **Apply**.

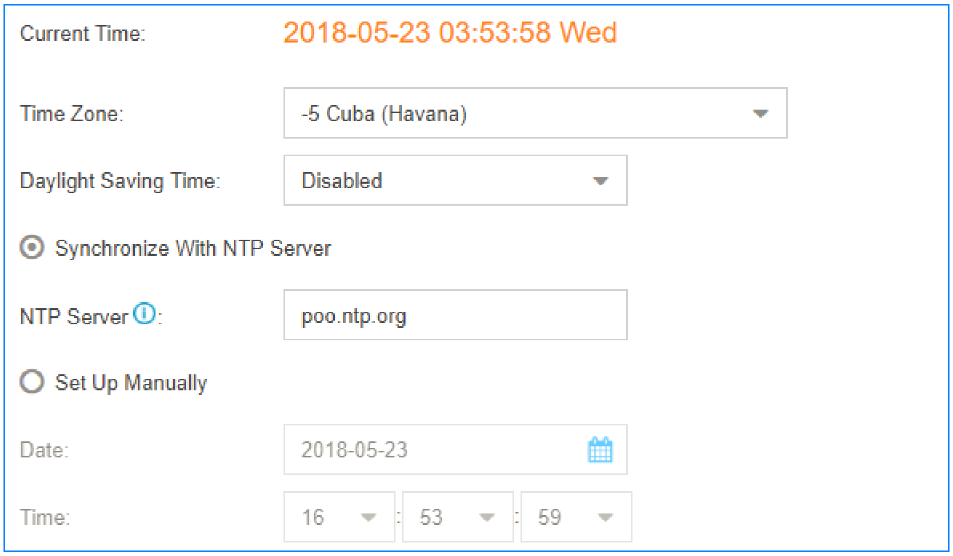
    Results: When the user logs in the PBX web interface by the extension user account, he/she can access the permitted configuration page.

Date and Time

To ensure that the time of logs and CDR is consistent with your local time , you need to adjust the date and time of the PBX.

On the **Date & Time** configuration page, you can see the current time of the PBX.

You can set the PBX time to be synchronized with a NTP server or set the time manually.



Change the PBX Time

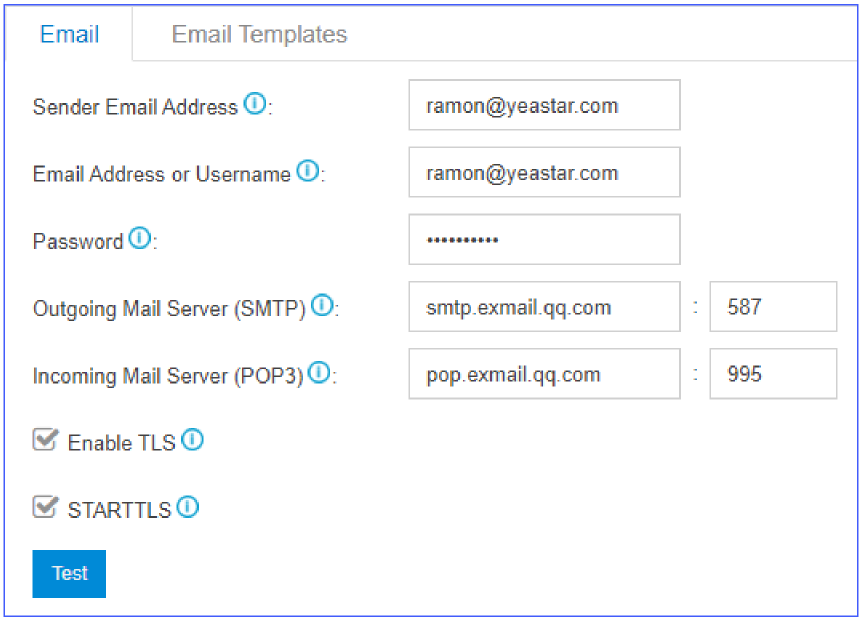
1. Go to **Settings > System > Date & Time**.
2. Select your current and correct **Time Zone**.
3. Check the option **Daylight Saving Time** if you need it in your place.
4. Click **Save**.
5. Reboot the PBX to take effect.

Email

The system email can be used to reset password, send voicemail to email, send alert event emails, and send fax to email. To make these features work, you need to set up the PBX system email.

Set up System Email

1. Go to **Settings > System > Email** to set up the system email.



- **Sender Email Address**: Enter an available email address.
- **Email Address or Username**: If the email server supports for User Name, enter user name. If not,enter the email address.
- **Password**: Enter the login password of the email address.
- **Outgoing Mail Server (SMTP)**: Enter the outgoing mail server and port according to the email server.
- **Incoming Mail Server (POP3)**: Enter the incoming mail server and port according to the email server.
- **Enable TLS**: Enable or disable TLS during transferring/submitting your Email to another SMTP server.

  Note: For Gmail or Exchange server, you need to enable TLS.

- **STARTTLS**: If you enable TLS, the STARTTLS is enabled by default . If the mail server doesn't support STARTTLS, do not select this option.

2. Click **Test** to check if the email works.
3. Click **Save** to save the email settings.

Auto Cleanup

Auto Cleanup is a feature that can auto clean your CDR, logs, voicemails, one-touch recordings periodically.

Table 3. Configuration Parameters of Auto Cleanup

CDR Auto Cleanup	
Max Number of CDR	Set the maximum number of CDR that should be retained. The old CDR will be deleted when the threshold is reached.

CDR Auto Cleanup	
CDR Preservation Duration	Set the maximum number of days that CDR should be retained.
Voicemail and One Touch Recording Auto Cleanup	
Max Number of Files	Set the maximum number of voicemail and one touch recording files that should be retained. The old CDR will be deleted when the threshold is reached.
Files Preservation Duration	Set the maximum number of minutes that voicemails and one touch recordings should be retained.
Logs Auto Cleanup	
Max Size of Total Logs	Limit the total size of pbxlog files in syslog. The old logs will be deleted when the threshold is reached.
Logs Preservation Duration	Set the maximum number of days that system logs should be retained respectively.
Max Number of Logs	Set the maximum number of event logs and operation logs that should be retained. The old logs will be deleted when the threshold is reached.

Event Center

You can set the PBX to send notifications when specific events or errors occur, notifying you via email.

For example, the system can automatically send a notification when the network connection is lost, VoIP trunk registration is failed, storage volume is running out of space, or the administrator password is changed.

Event Settings

Go to **Settings > Event Center > Event Settings** to configure the event settings.

- **Record**

  indicates that Record function is enabled. When the event occurs, the PBX will record the event in Event Log.

  indicates that Record function is disabled. When the event occurs, the PBX will NOT record the event in Event Log.

- **Notification**

indicates that Notification function is enabled. When the event occurs, the PBX will send notification to the Notification Contacts.

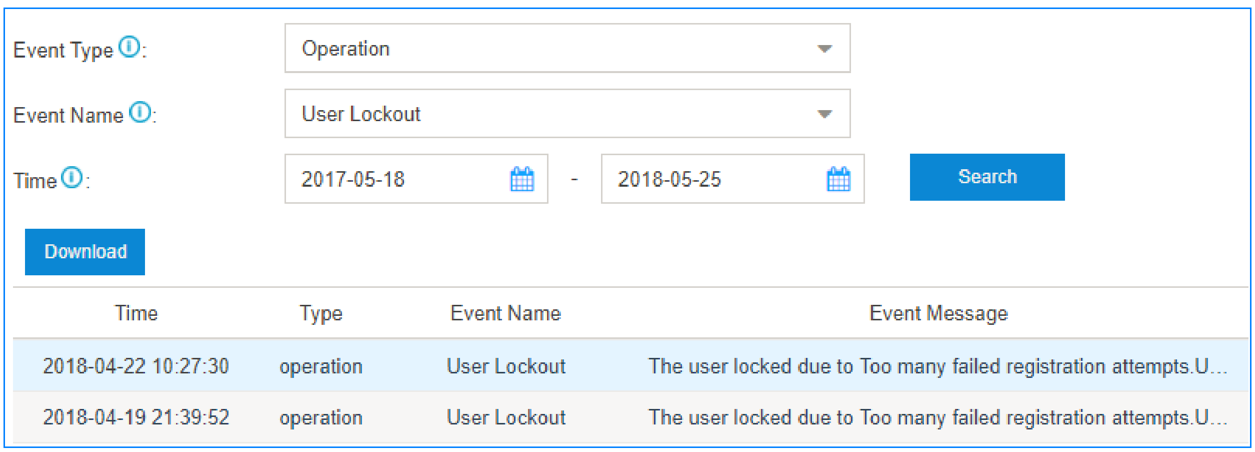
indicates that Notification function is disabled. When the event occurs, the PBX will NOT send notification to the Notification Contacts.

- **Edit Notification**

Click to edit the template of notification email.

Event Log

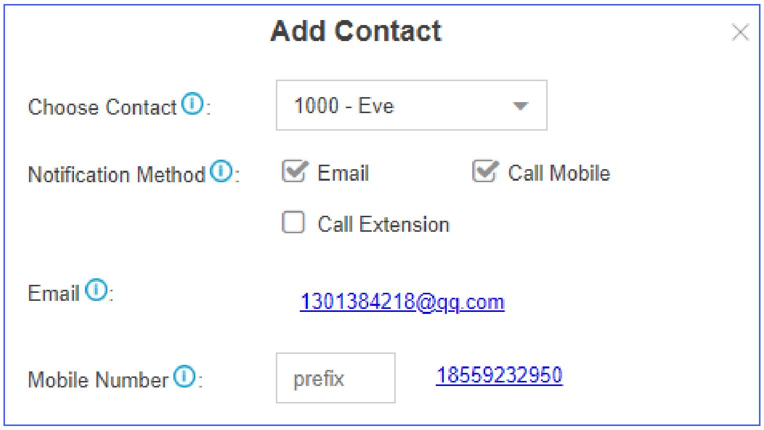
Go to **Settings > Event Center > Event Log** to search and check event logs.



Add Notification Contacts

You can set the PBX to send notifications when specific events or errors occur, notifying you via email.

1. Go to **Settings > Event Center > Notification Contacts**, click **Add**.
2. On the configuration page, choose a contact and set the notification method.



- **Choose Contact**: Choose an extension user or choose **Custom** to add an external contact.
- **Notification Method**: Select how to notify the contact when the event occurs.
  - **Email**: The PBX will send notifications to the email address of the contact.
  - **Call Extension**: The PBX will call the extension number of the contact when the event occurs.
  - **Call Mobile**: The PBX will call the mobile number of the contact when the event occurs.
- **Email**: If you choose **Notification Mode** to **Email**, you need to set the email address of the contact.

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

Remote Management

Yeastar Remote Management provides an affordable, low maintenance solution for easily deploying Yeastar VoIP PBX and VoIP gateways across multiple locations, reducing complexity and providing deep visibility and control.

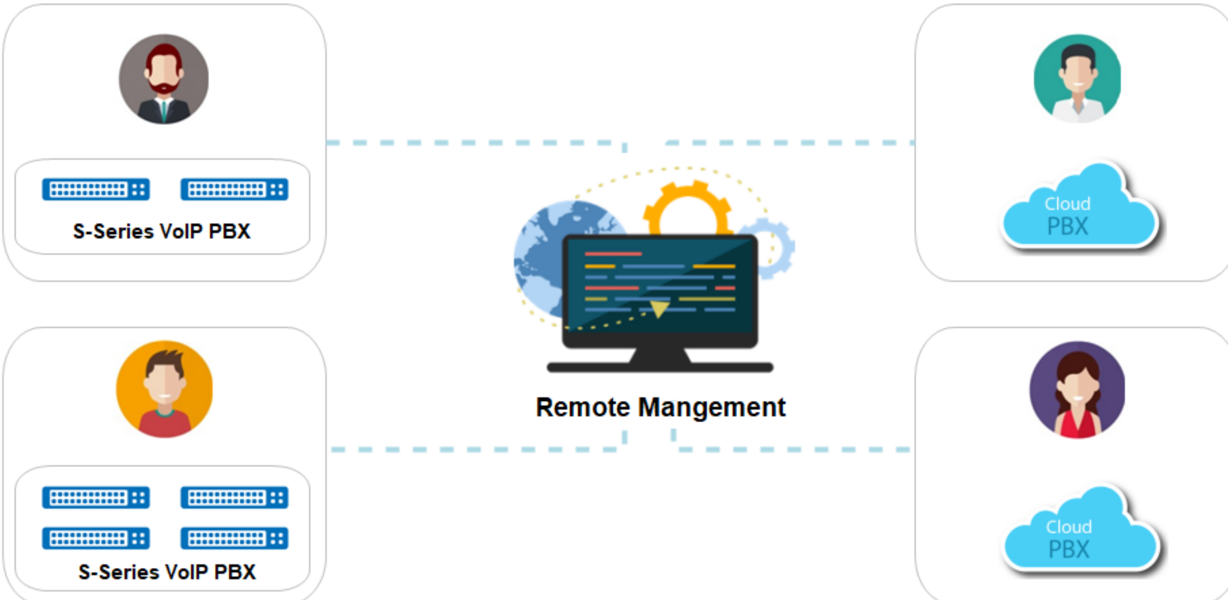
Compatibility

The following Yeastar products supports Remote Management feature:

- Yeastar Cloud PBX: 81.4.0.X or later
- Yeastar S-Series VoIP PBX: 30.6.0.20 or later
- Yeastar TA1600/TA2400/TA3200 V3

Remote Management Guide

How to manage Yeastar products on the Remote Management platform, refer to the Remote Management Guide.



API

Yeastar Cloud PBX provides API interfaces for you to integrate a third-party software or device.

Compatibility

API feature is supported on Yeastar Cloud PBX v81.4.0.8 or later.

API Guide

For more information of API, refer to [Yeastar Cloud PBX- API guide]().

In the API guide, we introduces how to enable and configure API on Yeastar Cloud PBX, and provides API references.

Maintenance

Maintenance gives you access to upgrade PBX firmware, check logs and troubleshooting.

Upgrade Firmware

Note:

- Back up the PBX configurations before you start to update the PBX firmware.
- If “Reset configuration to Factory Defaults” is enabled, the system will reset to factory default settings after upgrading.
- When update the firmware, please don’t turn off the power. Or the system will get damaged.

Related tasks
[Create a Backup File]

Upgrade Firmware

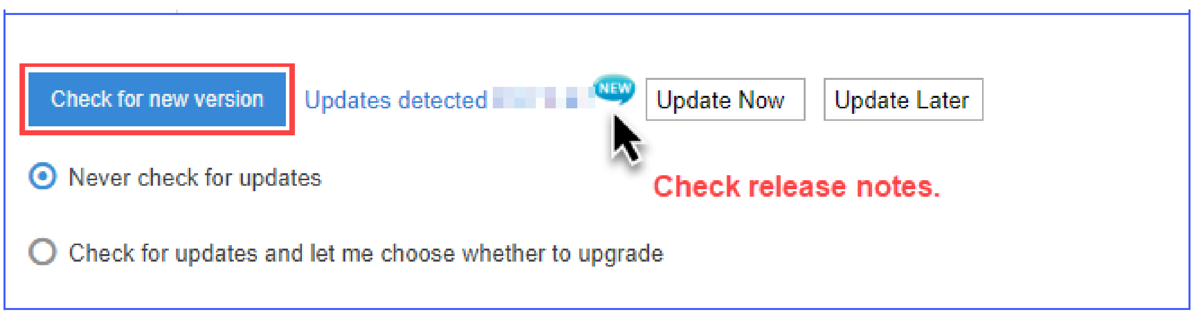
You can check for new version immediately or schedule automatic firmware check, if the PBX has a new released version, upgrade the PBX firmware with just one click.

Note: Make sure that the PBX can access the Internet, or the upgrade will fail.

Check firmware and upgrade immediately

1. Go to **Maintenance > Upgrade**.
2. Click **Check for new version** to check for new firmware immediately.

   If a new version is detected, you can click **New** check the release notes and decide whether to upgrade or not.



Schedule automatic update

1. Go to **Maintenance > Upgrade**.
2. Select one of the following options:

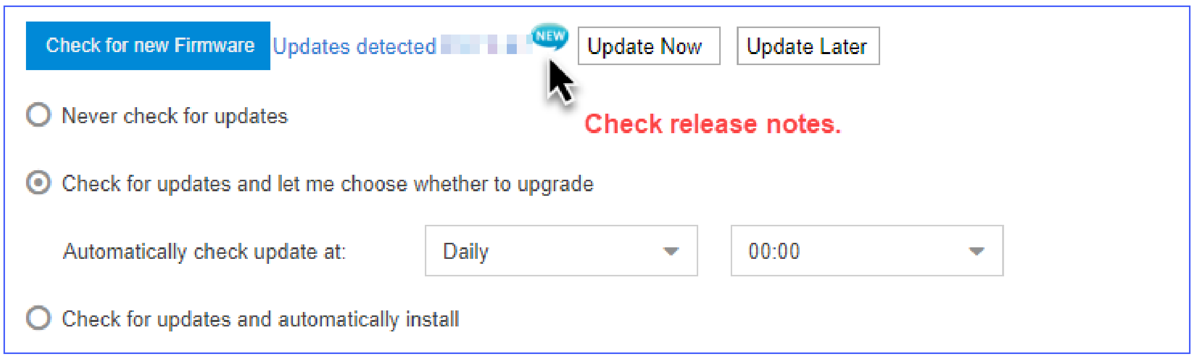
   - **Never check for updates**

     This option disables Automatic Updates.
   - **Check for updates and let me choose whether to upgrade**

This option notifies you that there are updates available. It requires user interaction to download them and install them.

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

   If a new version is detected, you can click **New** check the release notes and decide whether to upgrade or not.



Backup and Restore

Go to **Maintenance > Backup and restore**, then you can back up all configurations of PBX. Once backed up, back up file will be displayed in the list. You can upload backup file from local client to PBX, or you can choose from backup list and restore.

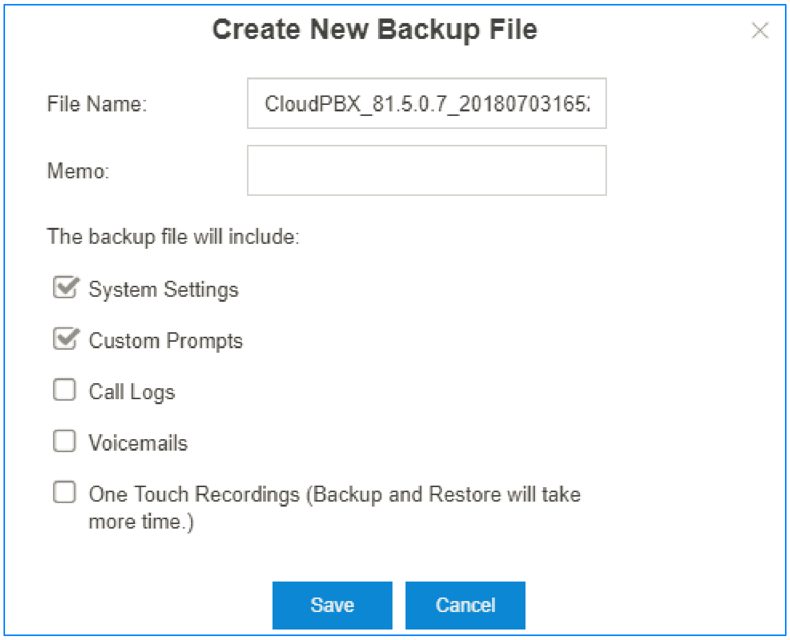
Create a Backup File

You can create a backup file of the PBX settings on the PBX web interface.

Note:

- For PBX version before 81.5.0.7, you can not back up one-touch recording files and voicemail files.
- For PBX version 81.5.0.7 or later, you can back up one-touch recording files and voicemail files.

1. Go to **Maintenance > Backup and Restore**, click **Backup**.



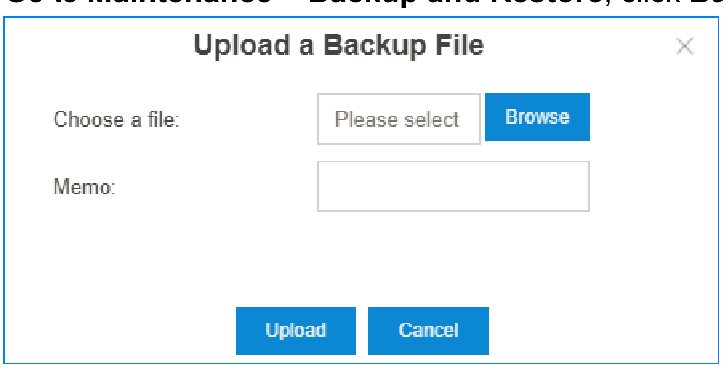
2. Set the **File Name**.
   The default file name contains the PBX model, firmware version, and backup date.
3. In the **Memo** field, enter notes for the backup file.
4. Select which configurations and files to back up.
5. Click **Save**.
   The created backup file will appear on the **Backup and Restore** page.

Upload a Backup File

You can select a backup file from your local PC, and upload the file to the PBX.

Note: The file format is `.bak` and the file name should not contain special characters.

1. Go to **Maintenance > Backup and Restore**, click **Backup**.



2. Click **Browse**, and select your backup file to upload.
3. In the **Memo** field, enter notes for the backup file.
4. Click **Upload**.
   The uploaded backup file will appear on the **Backup and Restore** page.

Restore a Backup File

After restore a backup file, the current configurations on your PBX will be OVERWRITTEN with the backup data.

Note:

- You cannot restore a backup file that is downloaded from a different PBX model.
- If a backup file is created from a newer version of PBX, you cannot restore this backup file. For example, restore a backup file (v30.7.0.35) to PBX (v30.6.0.16) would not work.
- You can restore a backup file that is created from a older version of PBX. For example, restore a backup file (v30.6.0.16) to PBX(v30.7.0.35) would work.

1. Go to **Maintenance > Backup and Restore**.
2. Choose a backup file, click .
   A pop-up window will appear at the bottom-right of the web page.
3. Click **Yes** to reboot the PBX.
   The PBX starts to restore data from the backup file.

Reboot the PBX

Reboot the PBX immediately on the PBX web interface or schedule auto reboot to keep the system running smoothly.

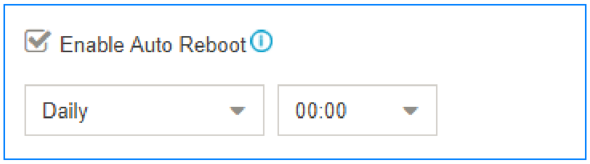
Note: When the PBX is rebooting, all the on-going calls will be terminated.

Reboot the PBX Immediately

1. Go to **Maintenance > Reboot**, click **Reboot**.

Schedule Auto Reboot

1. Go to **Maintenance > Reboot**, check the option **Enable Auto Reboot**.

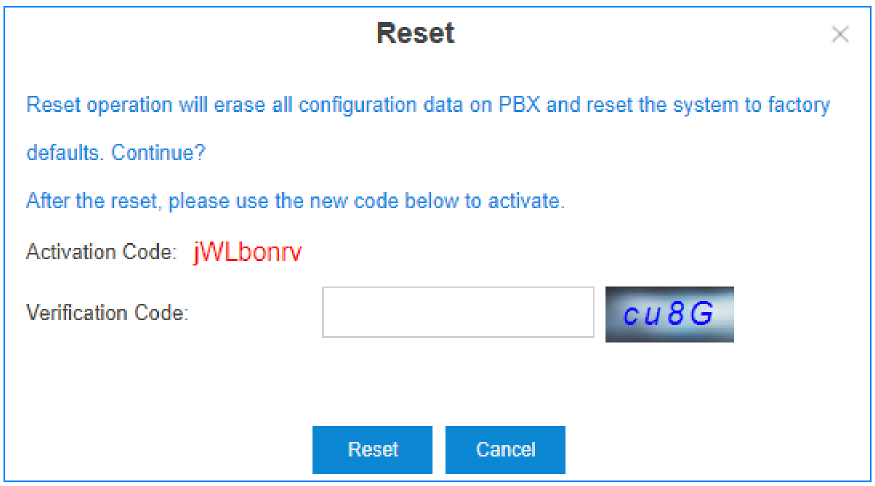


2. Set the frequency and time of auto reboot.
3. Click **Save**.

Reset the PBX

If you want to erase all the configurations on your PBX, you can reset the PBX to the factory defaults.

1. Go to **Maintenance > Reset**, click **Reset**.



2. Note down the new activation code.
   You need to use the new activation code to activate the PBX after reset.
3. Enter the verification code.
4. Click **Reset**.

System Log

The PBX automatically trace the PBX information, notices, warnings, errors, debug logs, and web logs, then generate log files. You can download the system logs on the PBX web interface, and check the logs.

Go to **Maintenance > System Log** to trace real-time logs or download the generated system logs.

System Log Settings

The PBX traces different levels of log.

- **Information**: Basic information.
- **Notice**: NOTICE information.
- **Warning**: WARNING information.
- **Error**: ERROR information.
- **DTMF**: DTMF information.
- **Time Log**: Add time stamp of system logs.
- **Debug**: Select the following checkboxes to decide which type of debug logs to trace:
    - **Enable SIP Debug**
    - **Enable RTP Debug**

System Log

The PBX generates system logs everyday. The system logs are compressed into a tar file. You can check the system logs on the **System Log** page.

Click to download the log file and open the log file by `Notepad++` or other editor software to check the logs.

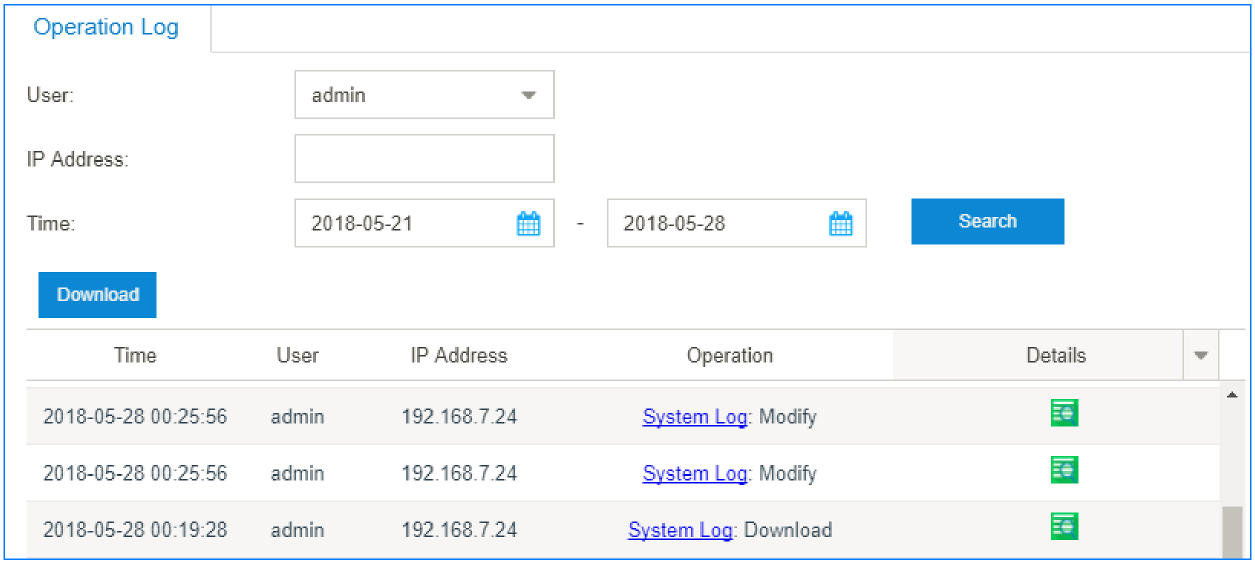
The PBX provides the following kinds of system logs:

- PBX firmware version
- AMI logs
- API logs
- Asterisk guard logs
- App logs
- Module update logs
- Linkus Cloud Service logs
- SSH connection logs
- PnP logs
- Web logs

Operation Log

The PBX records all the users' operations, and keep the logs in Operation Log.

Go to **Maintenance > Operation Log** to search and check the operation logs.



Troubleshooting

Yeastar Cloud PBXEthernet Capture Tool, IP Ping and Traceroute can be used to debug and capture packets.

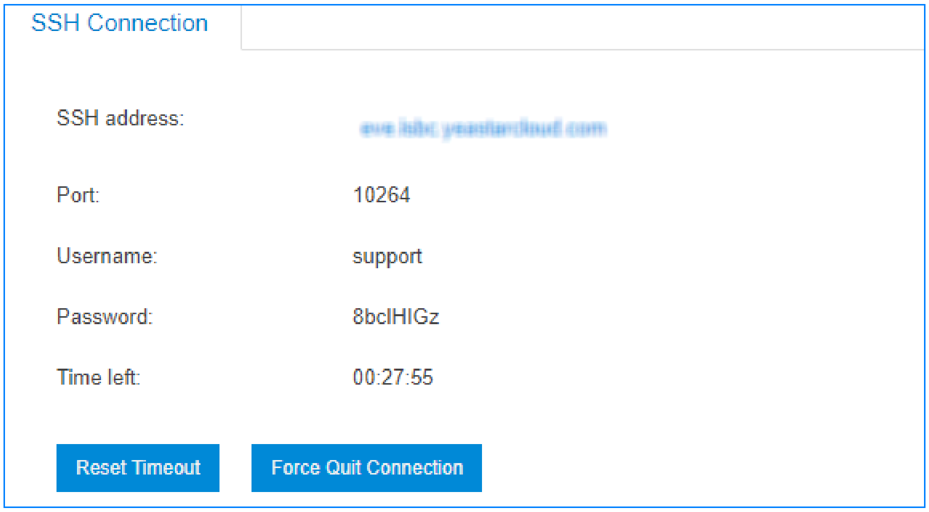
Access the PBX via SSH

To debug the system, you can establish a temporary SSH connection on the PBX, and access the PBX via SSH to check the logs.

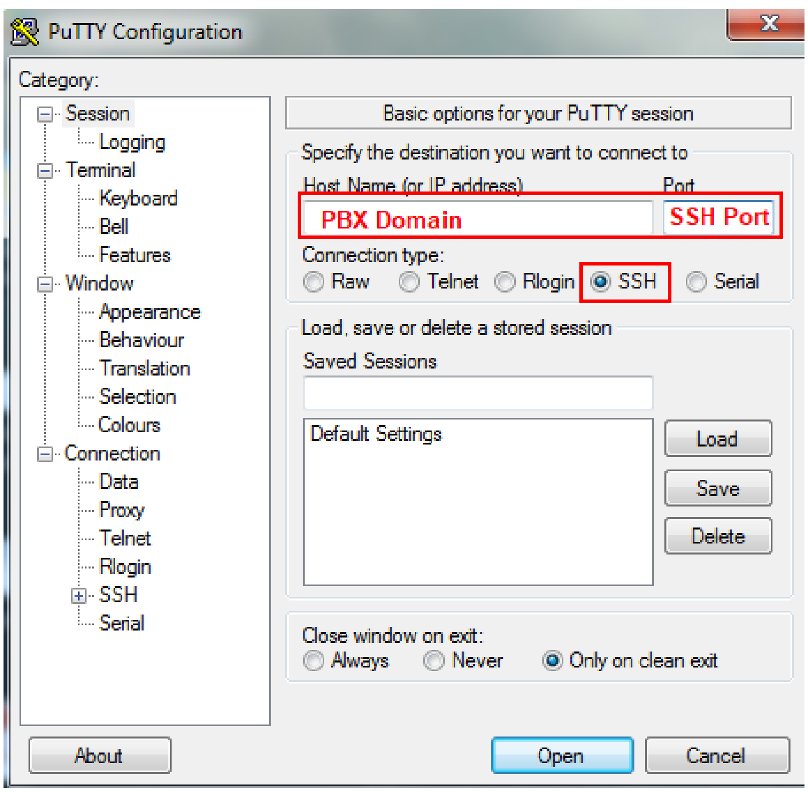
1. Establish SSH connection.
    a. Go to **Maintenance > SSH Connection**, click **Establish Connection**.
    b. On the **Time Settings** dialog box, select the time period in the **Timeout** drop-down menu, click **OK**.

       Note: When the connection times out, the SSH tunnel will be closed, you will not be able to access PBX via SSH.

       The SSH connection information is displayed on the **SSH Connection** page.

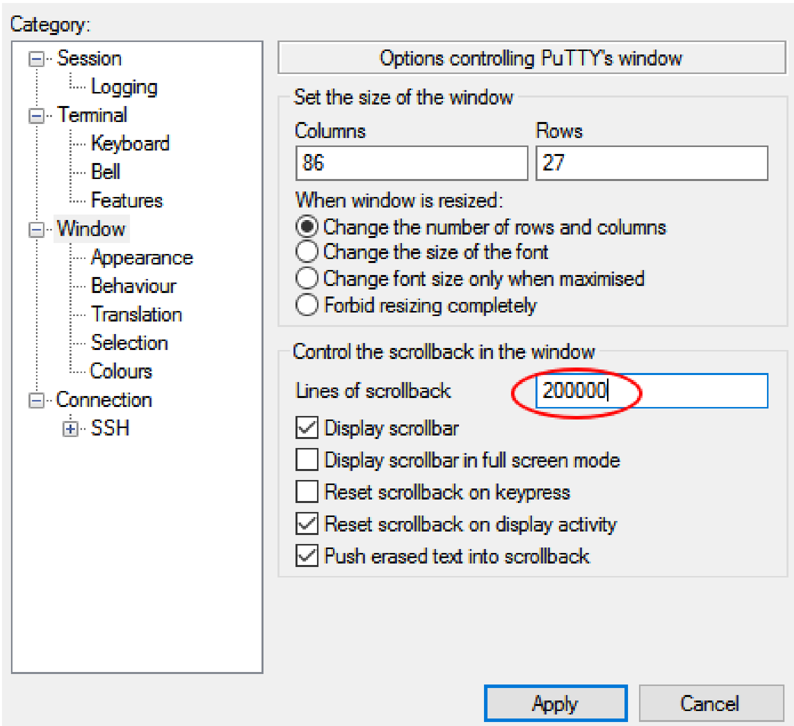


2. Use PuTTY to access the PBX via SSH.



- **Host Name (or IP address)**: Enter the PBX domain.
- **Port**: Enter the SSH port.
- **Connection Type**: Choose **SSH**.

3. To get more logs in the window, set the **Lines of scrollback** to a larger value, click **Apply**.



4. Enter the username and password to access the PBX.
    - **login as**: Enter `support`.
    - **password**: Enter the SSH password.

Tip: After copying the SSH password, right click on the Putty interface to paste password.

Capture Ethernet Packets

When there is a problem on the VoIP extensions or trunks, you can use the Ethernet Capture Tool to capture Ethernet packet, and download the packet to analyze it.

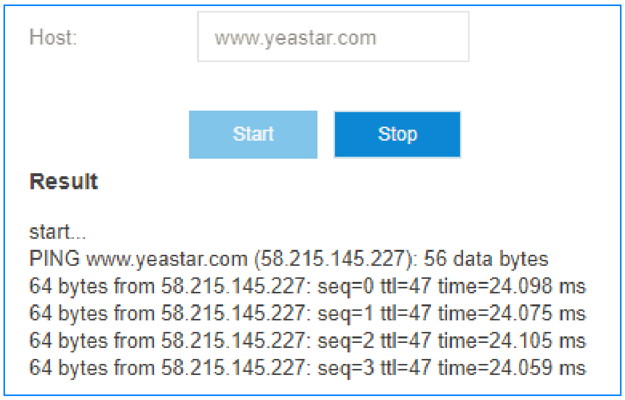
1. Go to **Maintenance > Troubleshooting > Ethernet Capture Tool**.
2. Click **Start**.
   The PBX will start to capture the Ethernet packet. During this time, you should duplicate the problem of your VoIP trunks or extensions.
3. Click **Stop** to stop capturing packets.
4. Click **Download** to download the captured packet.

   Decompress the `.tar`file and use `Wireshark` software to open the packet file.

Ping IP Address

A ping utility sends test messages from the local client to a remote target over the TCP/IP network connection. You can use IP Ping tool to test if the PBX can access the target IP address.

1. Go to **Maintenance > Troubleshooting > IP Ping**.

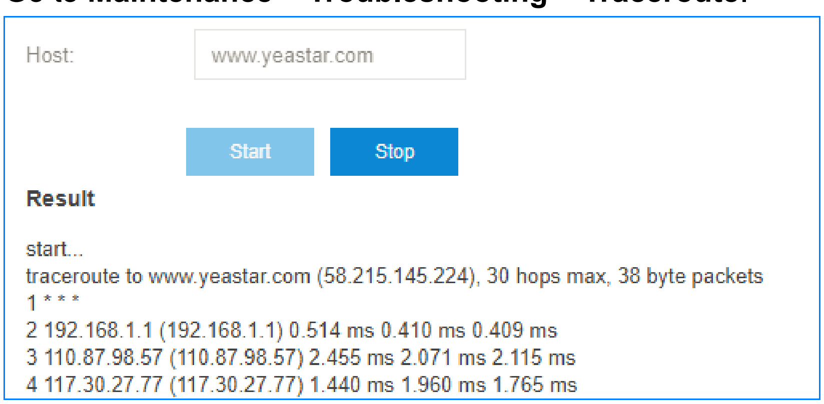


2. In the **Host** field, enter the target domain name or IP address.
3. Click **Start** and check the result.
4. Click **Stop** to stop ping.

Traceroute

Traceroute is a common diagnostic tool for displaying the route (path) and measuring transit delays of packets across a network.

1. Go to **Maintenance > Troubleshooting > Traceroute**.



2. In the **Host** field, enter the target domain name or IP address.
3. Click **Start** and check the result.

4. Click **Stop** to stop traceroute.

PBX Monitor

The PBX monitors the status of Trunks, Extensions, Concurrent Call, Conference.

You can log in the PBX web interface, go to **PBX Monitor** to check the real-time status of your trunks, extensions, and conferences.

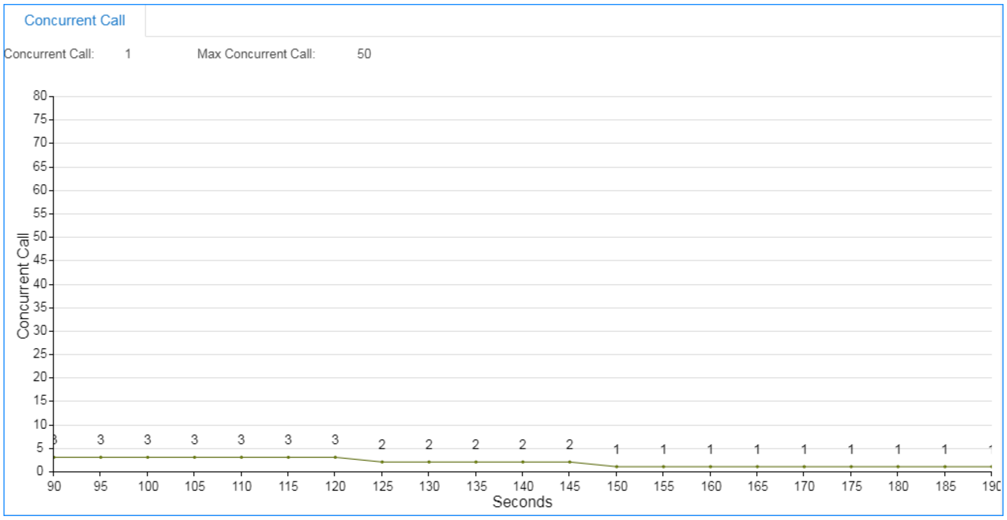
VoIP Trunk Status

Table 4.

Status	Description
	Registered
	Registering
	• Unreachable • Registration failed, caused by: ◦ wrong password ◦ wrong authentication name ◦ wrong user name ◦ transport type inconsistent

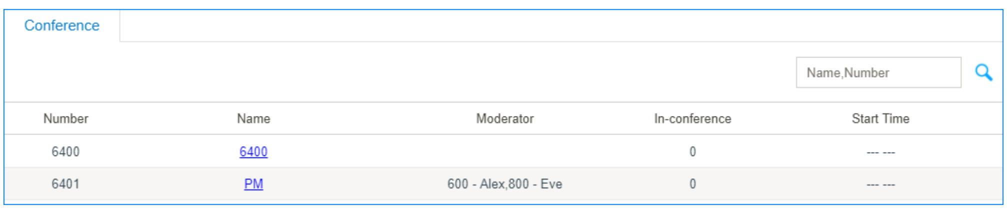
Concurrent Call

Check the maximum supported concurrent calls and the real-time concurrent calls on the PBX.



Monitor Conference

Check how many conferences are created on the PBX, and monitor the status of the conferences.

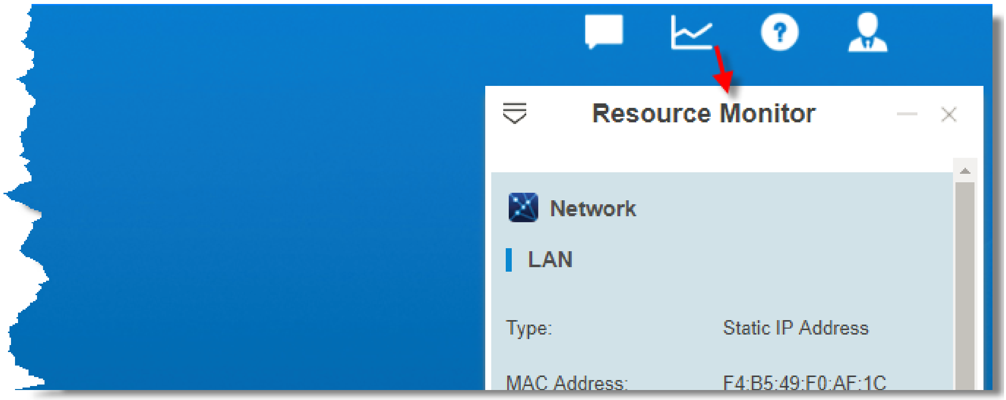


Number	Name	Moderator	In-conference	Start Time
6400	6400		0	--- ---
6401	PM	600 - Alex,800 - Eve	0	--- ---

Resource Monitor

Monitor the CPU usage, memory usage, disk utilization and network flow.

You can go to **Resource Monitor** to check the information or click the shortcut icon at the right-top corner.



Information

Check the basic information of the PBX.

- Product
- Serial Number
- Hardware Version
- Software Version
- System Time: The current time on the PBX.
- Uptime: The system up time since the last reboot.
- Extensions/Max Extensions: The number of added extensions/Maximum number of extensions allowed to be added

Network

Check the status of local network, cellular network, and VPN network.

Storage Usage

Check the usage of local storage in the PBX.

Recording Usage

CDR and Recordings

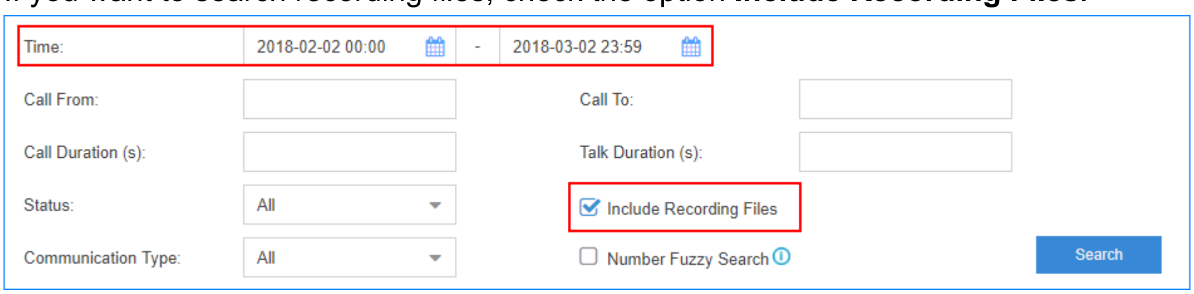
You can check CDR and auto recordings on the PBX web interface. CDR (Call Detail Record) is a data record that contains various attributes of the call, such as time, duration, call status, source number, and destination number, etc.

Searching Criteria

You can search CDR and recordings by the following criteria:

- **Time**: Set the start date and the end date to filter the call logs that are in the date duration.
- **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. Enter a value to filter the call logs that have call duration equal or greater than this value.
- **Talk Duration**: The time between the call answered and the call ended. Enter a value to filter the call logs that have talk duration equal or greater than this value.
- **Status**: Call status, including "answered", "no answered", "busy", "failed", and "has voicemail".
- **Communication Type**: Communication type, including "internal", "inbound", "outbound", "callback", "PBX warning call", "transfer", and "multisite interconnect".
- **Include Recording Files**: Check the option if you want to filter the calls that had been recorded.

Search CDR and Recordings

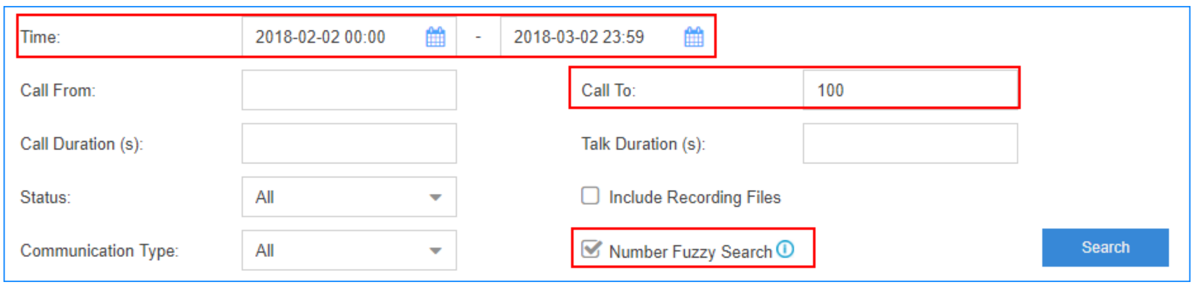
1. Log in the PBX web interface, go to **CDR & One Touch Recording**.
2. Set the **Time** to filter the call logs during the date duration.
3. If you want to search recording files, check the option **Include Recording Files**.


4. Set other searching criteria.
5. Click **Search**.
   The filtered call logs will display.

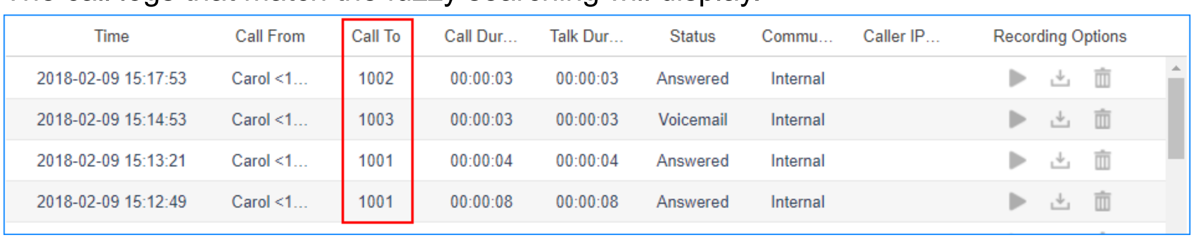
Fuzzy Search CDR and Recordings

By default, you need to enter an exact and complete phone number in the relevant searching criteria, or you cannot get the search result. If you cannot remember the exact number or the name, you can use Fuzzy Search feature.

1. Go to **CDR & One Touch Recording**.
2. Set the **Time** to filter the call logs during the date duration.
3. Enter a desired number or letters in **Call From** field or **Call To** field.
4. Check **Number Fuzzy Search**.



5. Set other searching criteria.
6. Click **Search**.
   The call logs that match the fuzzy searching will display.



Download CDR and Recordings

You can download the searched CDR or recording files to your local PC.

1. Go to **CDR & One Touch Recording**.
2. Search the CDR and Recordings.
3. To download the searched CDR, click **Download CDR**.
4. To download the searched recording files, click **Download Recordings**.