Tencent Real-Time Communication

Console Guide

Product Documentation

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Tencent RTC manages various businesses or projects through applications. Different applications can be individually created for distinct businesses or projects within the TRTC console, hence facilitating the isolation of data between businesses or projects.

Points of Attention

Each Tencent Cloud account has the capacity to establish a maximum of 300 Tencent RTC applications.

Creating an Application

1. Log in to the Tencent RTC Console and click **Create Application**.
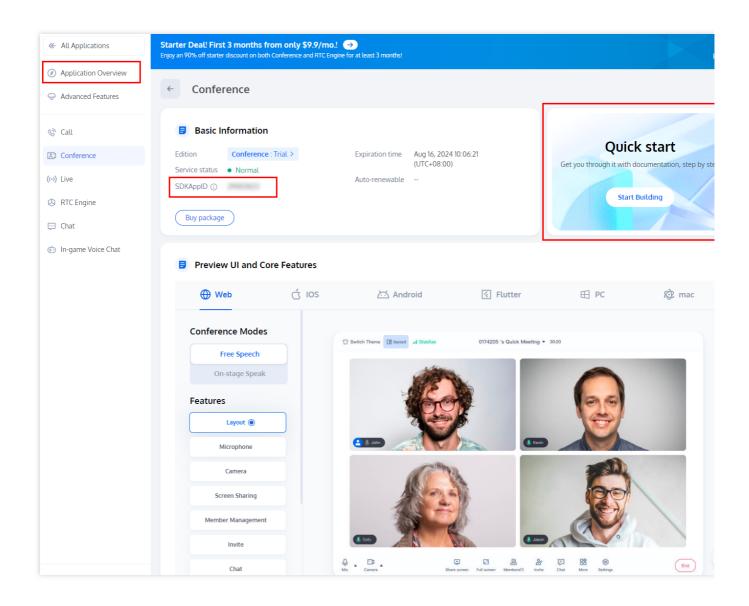
2. In the creation pop-up, based on the actual business needs, select a product and enter the application name, select the Data Storage **Region**, and click **Create**.
Note:
The default Data Storage for real-time audio and video service data is in Singapore, and the storage for instant messaging service data is in your selected data center.

3. After completing the application creation, you will automatically enter the application details page of the selected product. You may click on **Quick Start** to refer to the integration guide. You can view the `SDKAppID` and `SDKSecretKey` in the **Application Overview**, which will be used in subsequent steps.

Viewing Application List

After the application is successfully created, the created application information will be displayed in the application section of the overview. Click the application you want to view to display its basic information.

The basic information displayed includes application name, SDKAppID, application description, TAG, application status, key, application creation time, and service area. Additionally, you can verify whether advanced functions have been enabled and which products have been utilized.

Information Item	Description
Application Name	The custom application name designated during application creation.

SDKAppID	The SDKAppID, automatically generated after the creation of the application, serves as the unique identifier for the application. This parameter must be provided when invoking the VOICE API interface.
Application Description	You have the option to create a custom description for the application.
Application Version	By default, RTC engine are set to the basic version. You may upgrade to a higher version in order to unlock additional features based on your business needs. For more information, see RTC-Engine Packages.To use the Call application, you need to claim the Free Trial version. You can upgrade to a higher version based on your business needs to unlock corresponding value-added features. TRTC Call Monthly Packages
Tag	Tag values are established in Application Info and are utilized to distinguish and manage your various resources on Tencent Cloud. For instance, a corporation may have numerous departments, each with one or more TRTC applications. In this case, the corporation can mark departmental information by adding tags to TRTC applications.
Service Status	The current service status of the application comprises two states: **Normal and Disabled**. If the service status indicates "Disabled" and it was not manually suspended, please ascertain whether the package balance is zero, and if the Tencent Cloud Account has an overdue balance.
Key	Key information utilized for initializing the SDK configuration file.
Creation Time	The time at which the application was successfully established.

Related Documents

To examine the fundamental detailed insights of an application, kindly refer to Application Information.

Should you need to arrange or peruse the application function configuration details, please consult Function Configuration.

Overview

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Once your application has been successfully created, you can proceed to view the detailed information of the application through configuration management. The displayed information includes the application's basic details, Advanced features and so on.

Application Information

Basic Application Information

1. Navigate to the Tencent RTC Console > Applications to view the application list.
2. Select the application name you wish to modify.

3. Enter the application details page, and through the Application Overview tab's "Basic Information" module you can view the current application's basic information and version data.

Information Item	Description
Application Name	The name defined by the user at the time of application creation, which can be customised and altered.
Tags	The application supports setting TAGs to label certain business information. For more details, see More.
SDKAppID	The SDKAppID, automatically generated after the creation of the application, serves as the unique identifier for the application. This parameter must be provided when invoking the VOICE API interface.
SDKSecretKey	Information serving as keys for initiating the configuration file of the SDK.
Creation Time	The time when the application was successfully established.
Application Introduction	An editable application brief, customizable as per requirements.

Status	The application's status is shown and can be altered by selecting other options from the drop-down menu below.
Region	See selection of Service region

Alteration of Application Information

1. Within Application Management, select the application for information modification, and click on the management button in its right side action bar to access the application's detailed page.

2. In the Application Overview tab, examine the "Basic Information" module and click on the icon to the right.

3. Upon entering the application's information editing dialog box, you can change the **Application Name** and the **Descriptio**n. By clicking **Confirm**, the changes will successfully be saved.

Note：

The **Application Name** field supports only digits, letters, and underscores, and cannot exceed 15 characters.

The **Description** field supports only digits, letters, and underscores, and must not exceed 300 characters.

Application Version Information.

By default, RTC Engine applications are set to 'Free Trial'. Depending on your business needs, upgrading to a paid version will unlock additional features. For detailed information, please refer to Features and Billing Instructions.

Free Trial

Standard

Pro

Call applications require a 7-day 'Free Trial'. Upgrading to a paid version will unlock additional features according to your business needs. For detailed information, please refer to Features and Billing Instructions.

1-to-1 Call

Group Call

Information Item	Description
Application Version	The current application's monthly subscription plan version. By default, RTC Engine applications are set to 'Free Trial' and Call applications require a 7-day 'Free Trial'. Depending

	on your business needs, you can unlock additional features by upgrading to a paid version. For detailed information, refer to Features and Billing Instructions. **Version Details**: Displays the basic and value-added audio and video features of the current monthly subscription package, including the supported SDK platforms. **Obtain a Free Trial Version**: Each application can obtain trial eligibility for the free version up to 2 times (limited to 10 times per account), each time with a validity period of 7 days. The capabilities of the trial version are consistent with those of the flagship version. Please note: if you subscribe to a paid version during the validity period of the trial version, your current trial version will expire and be converted to the paid version you have subscribed to.
Expiration Time	The expiration time for the current application's monthly subscription version. You can manually purchase the same version to extend its validity period. If you need to switch to a different paid version, please submit a service ticket.
Auto-Renewal	Enables the auto-renewal feature upon expiration for the current application's monthly subscription package.

Application Service Status

The service status of the current application (SDKAppID) can be either normal or disabled. When the status is normal, the TRTC service is available. When it's disabled, the service is unavailable.
For application deactivation not initiated by you, this may be due to not enabling pay-as-you-go services or arrears. Please respond by enabling pay-as-you-go services, purchasing audio and video duration packages, or clearing arrears to reactivate the application. Doing so will ensure normal service availability.

TRTC Service Status

Primarily displays the status of the current application's TRTC basic and value-added services, including the states of "Normal" and "Disabled".
Enabled
When the status is normal, both the TRTC basic services and value-added services can be used as expected. To ensure uninterrupted service availability, please renew the package in a timely manner and ensure that your Tencent Cloud account maintains an adequate balance.

Disabled

When the status is disabled, both the TRTC basic services and value-added services are unavailable. Please check for the following scenarios:

Case 1: Self-disabled Application

1. Select the dropdown menu **More** and click on **Disable Application**.

2. Read the **<Disable Statement>** and confirm by clicking **Disable**.

3. Upon successful initiation of the disable operation, it takes around 3-5 minutes for the changes to be universally effected. Your patience is appreciated.

Self-enable Application

1. Click **Enable Application**.

2. Read the <Enable Statement> and click **Enable**.

3. Once the enabling operation is successfully initiated, it requires approximately 3 to 5 minutes for the changes to fully take effect. Kindly refresh the page later to view the updates.

Self-service Application Deletion

1. Select **More** and click **Delete Application**.

2. Read the <Delete Statement> and click Confirm **Delete**.

3. Once the deletion operation is initiated successfully, it takes approximately 3 to 5 minutes for the changes to take full effect. Please wait patiently.

Note：

Once deactivation/deletion takes effect, it only prevents new users from entering the room. Users who entered the room before the changes took effect will continue to be billed normally. You can use the Close Room API to forcefully shut down rooms that are currently in use.

Upon the effect of deactivation or deletion, the basic services (Call/RTC Engine) and value-added services (including mix stream transcoding, cloud recording, etc.) of the application will become unavailable.

Upon initiating the disable/enable/delete process for your application, please note that it takes approximately 3 to 5 minutes for the changes to fully take effect. We kindly ask for your patience during this time.

Under normal circumstances, it is not possible to delete an application directly. Please first disable the application on your own initiative, then proceed with the deletion.

Tags

Tags is used to identify and orchestrate your various resources on Tencent Cloud. For instance, a corporate entity may be divided into numerous business divisions, each incorporating one or more TRTC applications. In this context, enterprise can demarcate department information by allocating a TAG to the TRTC application.

Adding an Application Tag

1. In **Application Management**, you can view the **All Product Applications List** where you can observe the Tags information of each application.

2. Click on **Manage** under **Operation** to enter the application details page. Once there, click on the **Edit icon** located next to **TAG**.

3. Navigate to the TAG editing dialog box, select the **TAG Key** and **TAG Value** that have already been created in TAG Management.

Note:

If the current TAG does not meet your requirements, please go to TAG Management to create a new one.

Multiple TAGS can be added to an application. Simply click on + Add to create a new TAG configuration box.

4. Click **OK** to save. The console will refresh and display whether the modification was successful.

Deleting Application TAG

1. In Application Management, select the application for which the TAG information requires modification; click

Manage on the operation bar on its right to go to the application details page.

2. In the **Application Overview** tab, view the "TAG" module and click **Edit** on the right.

3. Enter the TAG editing dialog box, select the TAG you wish to delete, and click on the delete button on the right.

4. Click **OK** to save. The console will refresh and display whether the modification was successful.

Related Documents

For instructions on creating a new application, please refer to Creating Application.

If you need to configure or view the function configuration of an application, please see Function Configuration.

Function Configuration

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After an application is created, you can enable the Relay to CDN, On-Cloud Recording, and advanced rights control functions for your application through function configuration. The modification takes effect about 5 minutes after the modification.

Relay to CDN

Notice

TRTC service based on UDP transport protocol connects audio and video streams to CSS through protocol conversion, which is called "Relay to CDN".
The Relay to CDN is disabled by default. CSS must be enabled before the Relay to CDN function is enabled.
When Relay to CDN is used for CDN live viewing, CSS will charge related fees based on the downstream traffic/bandwidth generated by live viewing. For details, see CSS > Traffic Bandwidth Charging Description.
You can use Relay to CDN to push audio and video to CSS. If you use the recording capability of CSS to complete the recording, recording and file storage fees will be incurred. For details, see Billing of On-Cloud Recording and Recording Delivery.
If you live in a console to Relay to CDN forward domain name (xxxx.livepush.myqcloud.com) binding the recording, transcoding, pornographic identification, watermark, such as charging function template, the Relay to CDN would produce the template corresponding value added cost.

Enable the Relay to CDN function

1. Log in to the Tencent RTC Console > Applications, select the application whose function configuration you want to modify and click **Manage**.

2. Select **Advanced Features** from the project column on the left, then click the button to the right of **Global auto relay** within the **Relay to CDN** section.

3. In the dialog box that is displayed, read the risk description carefully. If yes, click **Save** to enable the Relay to CDN.

Switch the Relay to CDN domain name

1. Your current default domain name called xxxx.livepush.myqcloud.com/xxx.tlivecloud.com, and support to switch your domain name in CSS console. If you need to switch other domain names, please first go to the CSS console for binding.

Note：

After the default domain name is changed, the recording template, callback template, yellow authentication template, and watermark template bound to the current domain name will become invalid. Bind these templates to the new domain name and then complete the switchover. After the domain name switchover takes effect, the audio and video generated in the new room will be pushed based on the new domain name.

2. Go to the Tencent RTC Console > Applications, select the application whose function configuration you want to modify and click **Manage**. Select **Advanced Features** from the project column on the left, then click **View Details** on Relay to CDN, and switch the Relay to CDN domain name on the configuration page.

Disable Relay to CDN

To disable the Relay to CDN, perform the following steps:

1. Log in to the Tencent RTC Console > Applications, select the application whose function configuration you want to modify and click **Manage**.

2. Select **Advanced Features** from the project column on the left, then click the button to the right of **Global auto relay** within the **Relay to CDN** section.

3. In the **disable global auto relay** dialog box that is displayed, read the risk description carefully. If yes, click Confirm.

On-Cloud Recording Configuration

Notice

The On-Cloud Recording function uses the TRTC capabilities. The recording fee is charged by the TRTC side. For details about the charging rules, see the Cloud Recording Charging Description.

If a recorded file is stored on a VOD platform, the storage fee is charged based on the storage capacity of the recorded file. For details about the charging rules, see VOD video storage (daily settlement) price description or VOD video storage resource package price description.

If a recorded video file needs to be played or downloaded, the VOD service charges for video acceleration. The charges are based on downstream accelerated traffic. For details about the charging rules, see VOD Video Acceleration (daily settlement) price description or VOD video acceleration resource pack price description.

The On-Cloud Recording function is disabled by default. To enable the cloud recording function, you need to enable CSS and VOD first.

Global Auto-Recording

On-Cloud Recording provides an automatic recording method that does not require manual initiation and management of recording tasks. To use this method, you need to configure a global automatic recording template and enable global automatic recording. You do not need to develop or control the initiation and termination of recording tasks. Enable Global Automatic recording. After it takes effect for 5 to 10 minutes, the recording task will be started when the streamers in the TRTC room uplink the audio and video, and the recording task will be stopped when the streamers in the room check out and exceed the set waiting time. Before using the cloud recording function, please go to the console Application List Management > View All configurations to enable cloud recording, as shown below:

On-Cloud Recording supports recording the audio and video of each anchor in the room into an independent file (single-stream recording), and also supports mixing the audio and video of multiple anchors in the room into a file (mixed-stream recording). You can select the single stream and confluence modes for recording at the same time, and it is only valid for newly created rooms after being opened. This function does not take effect for rooms created before automatic recording is enabled. For a detailed implementation, see Achieve cloud recording and playback.

Global Single-Stream Recording

Global single-stream recording format supports audio and video recording, audio recording, video recording, recording files support MP4, HLS and AAC (audio recording format), for details about the recording file slicing policy, see Record file segmentation description.

Features	Description
Recording Mode	single-stream recording: The video footage of each anchor in the room is saved to a separate file.

	mixed-stream recording: The video footage of all the anchors in the room is mixed into one file for recording.
Recording Format	Audio and video format: Record audio and video streams in the room, suitable for video calls and interactive live broadcast scenarios. Audio only format: Record only the audio streams in the room.
File Format	Support MP4, HLS and AAC (audio only format).
Recording Duration of A Single File	The value which can be used to **specify the recording slicing duration** ranges from 1 minute to 1440 minutes. The default value is 1440 minutes.
The Duration of The Continuative Wait	If the interruption interval does not exceed the specified timeout period, only one file is generated for a call (or live broadcast), but the recorded file can be received only after the timeout period expires. The value ranges from 5 to 86400, in seconds. (Audio fee will be charged during the continuation period, please set the duration of the continuation reasonably.) Notice: In mixed-stream mode, black frames and silent frames continue to be recorded during the waiting period. In single-stream mode, black frames are not recorded. During the continuation waiting period, single-stream and mixed-stream recordings are charged for the duration of the audio.
Recorded file storage	Supports storage to VOD VOD: Supports the specified VOD application, the storage time of recorded files in the VOD, and the binding of the VOD task flow.

Note：

In single-stream recording mode, the audio and video streams in the room will be recorded separately according to the push parameters, without the need to set transcoding.

The recording robot will continue to wait for the anchor to go up in the room to complete the recording, and it will not end immediately after the anchor checks out. Please set it properly.

Note：

Single stream recording can record a maximum of 25 anchors in a room. If more than 25 anchors are recorded, the first 25 anchors will be recorded according to the check-in time. (If you need to record more than 25 anchors in a single stream, see API Manual recording.)

The default waiting time for continuation is 5s. If the time without anchor in the room exceeds the set continuation time, it will be divided into multiple recording tasks, and the corresponding recording files are also independent. If the time

without anchor in the room does not exceed the continuation time, the recording tasks will be maintained and the recording files will not be split.

Global Mixed-Stream Recording

In mixed-stream recording mode, you need to set additional video transcoding parameters, audio transcoding parameters and mixed-stream layout parameters of the automatic recording process to control the output specifications of the recorded file. The following table describes the parameters for global mixed-stream audio and video transcoding.

Features	Description
Video resolution	The video width and height are limited to [16,2560], and the video width * video height cannot exceed 2560 * 1440, the default video width is 640px, and the video height is 480px
Video bit rate	The video bitrate is limited to [64, 8192] and the default is 550kbps
Video frame rate	Video frame rate is limited to [1,60] and the default is 15fps
Video GOP	GOP is limited to [1,10] and defaults to 10s
Audio sampling rate	Supports 48000, 44100, and 16000Hz
Audio bit rate	The audio bit rate is limited to [32, 128] and the default is 64kbps
Audio channel	Supports single and dual channels

You can set prefabricated layout templates for global mixed-stream recording:

Grid layout: The videos of anchors are scaled and positioned automatically according to the total number of anchors in a room. Each video has the same size. Up to 25 videos can be displayed.

Screen sharing layout: The video of a specified anchor occupies a larger part of the canvas on the left side (if you do not specify an anchor, the left window will display the canvas background). The videos of other anchors are smaller and are positioned on the right side. If the total number of videos is 17 or less, the small videos are positioned from top to bottom in up to two columns on the right side, with eight videos per column at most. If there are more than 17 videos, the additional videos are positioned at the bottom of the canvas from left to right. Up to 24 videos can be displayed. A user who publishes only audio will still be displayed in one window.

Floating layout: By default, the video of the first anchor in the room (you can also specify an anchor) is scaled to fill the screen. When other anchors enter the room, their videos appear smaller and float over the large video from left to right starting from the bottom of the canvas. If the total number of videos is 17 or less, there will be four windows in each row (4 x 4); if it is greater than 17, there will be five windows in each row (5 x 5). Up to 25 videos can be displayed. A user who publishes only audio will still be displayed in one window.

Screen sharing layout and floating layout need to specify large screen user rules (the room needs to be laid out as a large screen anchor user ID prefix):

If the anchor ID in the room is not matched according to the prefix, the reserved layout will be left out for the screen sharing layout, and the floating layout will be arranged according to the anchor who entered the room the earliest.

If there are multiple anchor ids in the room, the layout of the anchor with the earliest check-in time will be followed.

Mixed-stream recording number: The number of people mixing under automatic recording is limited to 2-8 people, and the default is 2 people.

Manually Customize the Recording

If you only need to use certain rooms to initiate recording tasks, you can do so manually through CreateCloudRecording.

Disable the On-Cloud Recording

After the on-cloud recording function is enabled, 「Manual Custom Recording」 is enabled by default and cannot be disabled. 「Manual Custom Recording」 generates recording usage based on your initiation. If you only need to use 「Manual Custom Recording」, disable 「Global Automatic Recording」.

Recording File Management

After recording, you can view the recorded file on the VOD audio and video management interface. We will upload the file to the storage location you specify, and you will be able to view it once the upload is complete.

By filtering the VOD applications that you specify, you can view the corresponding recording result files.

By clicking a file name, you can view detailed file information, including preview videos, obtain file urls, and download files.

Note：

Depending on the size of the recorded file, the upload time may be delayed by 30 seconds to several minutes. You can subscribe to the recording callback event to obtain the successful upload time of the file.

Advanced Permission Control

If you want to add room access restrictions or access restrictions to some rooms, that is, only allow specified users to enter the room or access the microphone, and you are worried that the client judge the permission is easy to encounter cracking attacks, then you can consider enabling advanced permission control.

Notice

After advanced permission control is enabled, all users of the current SDKAppID need to pass the privateMapKey parameter correctly in TRTCParams in order to successfully enter the room. If there is a user using this SDKAppID

online, do not enable this feature.

Enable advanced permission control

1. Log in to the Tencent RTC Console > Applications, select the application whose function configuration you want to modify and click **Manage**. Select **Advanced Features** from the project column on the left.

2. In **Advanced Permission Control**, click the **Enable** Advanced Permission Control button on the right, click **Confirm** within the pop-up window to complete the activation..

Disable advanced permission control

1. Log in to the Tencent RTC Console > Applications, select the application whose function configuration you want to modify and click **Manage**. Select **Advanced Features** from the project column on the left.

2. In **Advanced Permission Control**, click the **Enable** Advanced Permission Control button on the right until it becomes closed.

Related Document

To create an application, see Creating an Application.

To view the basic information about an application, see Application Information.

Callback Configuration

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The Event callback service notifys your server of real-time audio and video service events in the form of HTTP/HTTPS requests. The Event callback service integrates some events in the Room Event and Media Event groups. You can fill in the callback configuration information on the Tencent RTC console by following the instructions below. After the configuration is complete, you can receive the callback event notification.

Preconditions

1. Go to the Tencent RTC Console > Applications. click on **Manage** in the row of the target application whose callback configuration needs to be modified, to enter the **Configuration** page.
2. Select **Advanced Features** from the project sidebar on the left, and within **Callbacks** configuration, you can set up the callback configuration details.

Set the callback key and URL

1. Click in the upper right corner of the callback Configuration TAB to **edit**.
2. Set the callback key (optional) based on service requirements.
3. Enter a callback URL based on service requirements (required):

Room event callback: Event notification for creating/dissolving rooms, entering/exiting rooms, etc. More callback event descriptions.

Media event callback: Start/stop push video data, start/stop push audio data, start/stop push auxiliary route data and other event notifications. More callback event descriptions.

Recording callback: It is used to call back events related to the on-cloud recording function, and supports event notification of the start and exit of the on-cloud recording module, the start of the cloud recording upload module, and the end of the upload. More callback event descriptions.

Relay to CDN callback: Event callbacks used to enable the Relay to CDN function. More callback event descriptions.

Note：

Callback URL Protocol header: HTTP, HTTPS, and can contain only the following characters: a-z, A-Z, 0-9, -, _, and?
, %, =, #,., /, and + contain a maximum of 2083 characters.
4. Click **Confirm** to successfully set the callback URL.

Package Management

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TRTC supports viewing the usage of Package under your account through the Console. You can click on the Left Sidebar Package Management and view the Package information in the List.

Package list

Note：

Package types are divided into Free Trial and monthly package, the former can deduct audio and video, recording, mixtranscoding time, the latter can deduct audio and video time, the specific deduction ratio details can be seen free time description.

Features	Description
Package Name	The purchased package name is supported as follows: RTC Engine - Standard RTC Engine - Pro TRTC Free Trial 1-to-1 Call Group Call
Package ID	The system automatically generates a unique package ID after a successful purchase

Package Type	There are two types of free time package and monthly package, among which: Standard/Professional packages are customer purchased products, including the Call/RTC Engine The Free Trial is a platform giveaway product, distributed for 12 months at a time, 10,000 minutes per month
Package Duration(min)	Some minutes are included in the Package
Remainging time of this cycle(minutes)/(%)	With the number of minutes remaining and the proportion of usage, you can see the usage of the package more intuitively
Start time of this cycle	The time the package was successfully purchased
End time of this cycle	Package expiration time
Status of this cycle	Refunded, Expired, Run out, Not in Use, In Use

Purchase package

If the actual service needs to renew the package, you can go to the TRTC application list, click Management to go to the application details page, and click Buy Official Package at the bottom of the page to purchase.

Note：

Before purchasing a package, make sure you understand the billing information. For details, see Billing Overview.

Usage Statistics
Interactive Audio/Video

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Real-time audio and video interaction statistics of the TRTC console, support to view the detailed consumption data of created applications.

Note

The data is updated every 5 minutes instead of in real time, so there may be a delay of 5-180 minutes in the display of statistics.

Directions

1. Go to the Tencent RTC Console and select **Usage** in the left column.

2. Select the application that you want to view, and select a time period for viewing.

3. The trend chart can be filtered to see paid and free usage.

4. Free usage: When on-cloud recording/relay to CDN is carried out in the room, the corresponding recording/transcoding robot will join the room as a virtual audience, subscribe to the audio and video stream of the anchor side, and generate the audio and video duration consumption. When the recording is stored to VOD/transferred to CSS, the robot usage is not charged.

Usage Details

Durations are calculated in seconds and rounded up to minute. Therefore, the cumulative duration used for billing may differ slightly from your actual usage. For the actual billable amount, see the bills generated in Billing Center.

Details (Jul 27, 2024 to Aug 02, 2024)

In the detailed records, durations are calculated by seconds and rounded to the nearest minute. A duration less than 1 minute is counted as 1 minute. Therefore, if you add up the minutes displayed in each line below, the sum will be slightly different from the actual minutes for billing. The final billing usage is up to the bill in Billing Center ⤢.

Time ⇕	Audio (min)	HD (min)	FHD (min)	2K (min)	2K+ (min)
			No data found		

On-Cloud Recording

Last updated：2025-02-25 16:47:48

On-Cloud Recording usage statistics of the TRTC console are used to display details on the cloud recording usage of TRTC by recording duration. If you are using the recording concurrent peak route billing for CSS, you need to go to the CSS console to check.

Note

The data is updated every 5 minutes instead of in real-time, so there may be a delay of 5-180 minutes in the display of statistics.

Directions

1. Log in to the Tencent RTC Console, and select **Usage** from the left sidebar > choose **On-Cloud Recording** on the page.

2. Select the application whose usage details you want to view, and select a time range.

Usage Details

Durations are calculated in seconds and rounded up to minutes. Therefore, the cumulative duration used for billing may differ slightly from your actual usage. For the actual billable amount, see the bills generated in Billing Center.

Details (Jul 27, 2024 to Aug 02, 2024)

In the detailed records, durations are calculated by seconds and rounded to the nearest minute. A duration less than 1 minute is counted as 1 minute. Therefore, if you add up the minutes displayed in each line below, the sum will be slightly different from the actual minutes for billing. The final billing usage is up to the bill in Billing Center 🗗.

Time ↕	Audio (min)	HD (min)	FHD (min)	2K (min)	2K+ (min)
			No data found		

On-Cloud MixTranscoding

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On-Cloud MixTranscoding consumption statistics of TRTC console are mainly used to show the consumption details generated by MCU cloud mixtranscoding provided by TRTC. If you use the mixtranscoding function billing provided by CSS, please go to the CSS console to check.

Notes

The data is updated every 5 minutes instead of in real-time, so there may be a delay of 5-180 minutes in the display of statistics.

Directions

1. Log in to the Tencent RTC Console, and select **Usage** from the left sidebar > choose **On-Cloud MixTranscoding** on the page.

2. Select the target application and time period (taking bill-by-duration mode as an example).

Usage Details

Durations are calculated in seconds and rounded up to minutes. Therefore, the cumulative duration used for billing may differ slightly from your actual usage. For the actual billable amount, see the bills generated in Billing Center.

Details (Jul 27, 2024 to Aug 02, 2024)

In the detailed records, durations are calculated by seconds and rounded to the nearest minute. A duration less than 1 minute is counted as 1 minute. Therefore, if you add up the minutes displayed in each line below, the sum will be slightly different from the actual minutes for billing. The final billing usage is up to the bill in Billing Center ⬀.

Time ⇅	Audio (min)	HD (min)	FHD (min)	2K (min)	2K+ (min)
			No data found		

Relay to CDN

Last updated：2025-02-25 16:47:48

The statistics of the TRTC console's Relay to CDN usage are mainly used to display the details of the bypass push service provided by TRTC. If you use the third-party repush billing provided by the CSS, go to the CSS console to view the statistics.

Notes

The data is updated every 5 minutes instead of in real time, so there may be a delay of 5-180 minutes in the display of statistics.

Directions

1. Go to the Tencent RTC Console, select **Usage** in the left column, and select **Relay to CDN** on the page.

2. Select the application you want to view, and select a time range (using the bandwidth charging mode as an example).

Usage Details

The detailed records show peak bandwidth in five-minute intervals or daily peaks, with monthly peaks used for billing. Estimate your monthly costs by examining the detailed logs, but refer to the Billing Center for the actual charges.

Details(To Tencent Cloud) (Jul 27, 2024 to Aug 02, 2024)

Peak bandwidths every 5 minutes or daily peak bandwidths are displayed in the detailed records, while monthly peak bandwidth is used for billing. You can estimate the monthly cost by checking the detailed records, and the final billing usage is up to the bill in the Billing Center ⤢.

Time ⇕	Bandwidth (Mbps)
No data found	

Monitoring Dashboard
Call Data
Call List

Last updated：2025-02-25 16:47:48

Tencent RTC offers a monitoring dashboard for developers to monitor call quality. You can view call details and learn about the call status of users in the monitoring dashboard.

Viewing Call List

Log in to the Tencent RTC Console, and click Monitoring Dashboard to view the call details of all rooms under the logged-in account.

The call list in the monitoring dashboard, displays 10 call records on one page by default, sorted in reverse order by the start time of the call, i.e. the first one is the most recently created call record. Each record includes 6 fields: Room ID, Start Time - End Time, Call Status, Room Duration, Joined Users, Operation.

The information displayed in the call list includes:

Item	Description
Room ID	The room ID of the call
Start Time - End Time	The start and end time of the call
Call Status	Includes two states: Not completed and Ended.
Room Duration	The time between the entry of the first user and exit of the last user. If a call is in progress, the time from the entry of the first user to when the query takes place is displayed.
Joined Users	The total number of users who have entered the room
Operation	Click Call Details to go to the details page and view the details of the call.
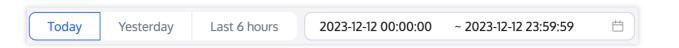
Searching for Calls

In the monitoring dashboard, you can filter the call list using a number of search methods.

To view the information of call rooms of a particular application, click **Select application**, select the application from the drop-down list, and click **Search**.

To view the information of calls during a specific time period, select **Today**, **Yesterday**, **Last 6 hours**, or enter a time range, and click **Search**.

To view the information of a particular call, select a time range, enter a room ID (`roomid`) or user ID (`userid`), and click **Search**.

| Enter Room ID | Enter User ID | All ⌄ | Confirm |

More Operations

If you want to view the call details of a particular room, see Call Details.

| Enter Room ID | Enter User ID | All ⌄ | Confirm |

Call Details

Last updated：2025-02-25 16:47:49

This document describes the information you can view on the details page of a particular call room in the monitoring dashboard.

Directions

1. Log in to the Tencent RTC Console, click Monitoring Dashboard, and find the room whose call details you want to view.

2. Click the **Room ID** or **View Call Details** on the right to go to the details page.

Information Displayed

Basic information

Call room information: the basic information of the call room, including **SDKAppID**, **Application Name**, **Room ID**, **User Count**, **Start Time - End Time**, and **Room Duration**.
Current time range: the time range queried. The calls of up to 5 hours can be displayed per query.

User list

This section shows the user information of the call, including user ID, role, type, region, stay in the selected time range, time of room entry and exit, length of stay, SDK version, SDK type, as well as device and network information.

Item	Description
User ID	By default, the recently joined 6 users are displayed. To display more than 6 users, click **Add user** or **View All Users**. Up to 20 users can be displayed.
User Role	Anchor or audience. Anchors can send and receive data, but audience can only receive data.
User Type	Upstream or downstream. A green up arrow means the user has sent data, and a yellow down arrow means the user has received data.
Region	The geographic location of the user
User's Stay in This Period	A blue segment represents the period when the user is in the room, and the gray segment represents the period when the user is not in the room.
Room In/Out Time	The time of the user's first entry and last exit.
Duration	The length of the user's stay in the room.
SDK Version	The version number of the user's SDK.
SDK Type	The type of SDK used by the user, see the documentation SDK Download for descriptions.
Device	The model or number of the device used by the user
Network	The network type used by the user

Data received and sent

A user in a call receives the data of other users while sending local data. It is therefore necessary to display the information of data going both ways. By default, the call details page of the monitoring dashboard displays statistics from the perspective of the receiver. You can also switch to the perspective of the sender.

Receiver

You can switch between four tabs: **All**, **Video**, **Audio**, and **Screen Sharing**. A tab is displayed only if the corresponding data is received.

By default, the graph shows the data received from all remote users, which is marked by different colors. You can view an overview of a single user's data by clicking **Select sender to view details** and selecting the specified sender user.

Red lines in the graph indicate network jitter. You can click **Select sender to view details** in the top right and select the corresponding user ID to check the details in End-to-End Details.

If a long time span is displayed, you can use the mouse wheel to zoom in part of the graph to view the details of a specific point.

Sender

You can switch between four tabs: **All**, **Video**, **Audio**, and **Screen Sharing**. A tab is displayed only if the corresponding data is sent.

Red lines in the graph indicate network jitter. You can click **View details** in the top right to check the details in End-to-End Details.

If a long time span is displayed, you can use the mouse wheel to zoom in part of the graph to view the details of a specific point.

Receiver | Sender

User_B
SDK version4.15.15.1518　Platform：Mac OS X/Safari 16.6()　　View Details

All | Video | Audio

Bitrate (Kbps)

100
50
0
50
100
14:06:30 14:15:00　14:30:00　14:45:00　15:00:00　15:12:13

User_A
SDK version4.15.15.1518　Platform：Mac OS X/Safari 16.6()　　View

All | Video | Audio

Bitrate (Kbps)

900
800
600
400
200
0
100
14:15:00　14:30:00　14:45:00　15:00:00　15:12:05

End-to-End Details

Last updated：2025-02-25 16:47:49

End-to-end call details are the details of data transferred across the sender-receiver link. The preconditions of a smooth audio/video call are good network connections and stable device performance, which are what you should start with when checking the end-to-end details of a call.

Directions

1. Log in to the Tencent RTC Console click Monitoring Dashboard, and find the room whose call details you want to view.

2. Click the room ID or **View Call Details** on the right to go to the Call Details page.

3. On the receiver/sender page, select the ID of the user whose data you want to view, and go to the end-to-end details page via either of two methods.

On the **Receiver** page, click **Select sender to view details** on the right and select a user ID.

On the **Sender** page, click **View Details**.
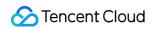
Information Displayed

The end-to-end details page shows the data of **Video**, **Audio**, and **Screen Sharing**, which can be viewed from the perspective of the receiver or sender.

Entering from the receiver page, the end-to-end details are the call details information **between the sender end and the receiver**, including **Basic Information** and the detail information of **Sender** and **Receiver** under the video and audio indicators.

Entering from the sender page, the end-to-end details are the call details information of the **sender**, including **Basic Information** and the details of **Sender** under the video and audio indicators.

Basic Information

At the top of the page, 6 fields of basic information are displayed, including SDKAppID, Application name, Room ID, User Count, Start time - End time, and Room duration.

SDKAppID	Application name	Room ID	User Count	Start time - End time	Room duratio
	Call_231211	17416033	2	2023-12-12 14:06:43～2023-12-12 15:19:01	1hour12min18s

Sender

Video Details

Video details include 4 major segments, video upstream and network packet loss, control metrics, video frame rate, and video send resolution, with multiple metrics in each segment presented as multi-dimensional visual charts.
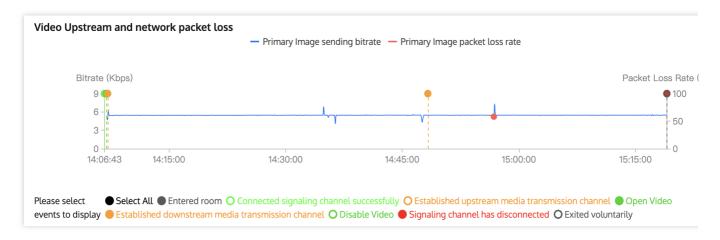
Video Upstream and Network Packet Loss

The image horizontal axis is the time granularity, whose size can be changed by sliding the scroll wheel; the left vertical axis is the **primary image sending bit rate** (blue line); the right vertical axis is the **primary image packet loss rate** (red line).

Different colored dots in the image indicate different types of events, you can filter and view different types of events by clicking the corresponding colored buttons below the image.

8 event types: Entered Room, Connected Signaling Channel Successfully, Establishing Upstream Media Transmission Channel, Open Video, Established Downstream Media Transmission Channel, Disable Video, Signal Channel Disconnected, and Exited Voluntarily.

Note: When entering end-to-end details from the sender, there are 7 event types, except for "Established downstream media transmission channel".
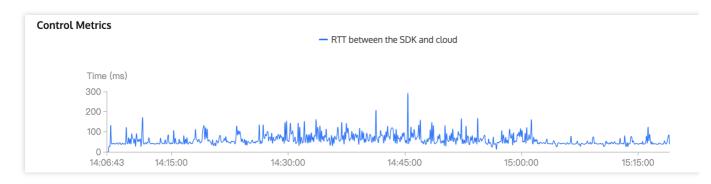

Control Metrics

The horizontal axis of the image is the time granularity, whose size can be changed by sliding the wheel.

The vertical axis is the **round-trip time (RTT) between the SDK and cloud** in milliseconds, which is represented by the blue line.

Video Frame Rate

The horizontal axis of the image is the time granularity, whose size can be changed by sliding the wheel;

The vertical axis is the frame rate. Hover the mouse over the image to view three metrics: **primary image capturing frame rate**, **primary image sending frame rate**, and **primary image RTC video encoding frame rate**.

Video Send Resolution

The horizontal axis of the image is the time granularity, whose size can be changed by sliding the wheel.

The vertical axis is the resolution and the primary image resolution can be viewed by hovering the mouse over the image.

Audio Details

Audio details include 3 major segments, audio upstream and network packet loss, control metrics, and audio capturing volume, with multiple metrics in each segment presented as multi-dimensional visual charts.

Audio Upstream and Network Packet Loss

The horizontal axis of the image is the time granularity, whose size can be changed by sliding the scroll wheel; the left vertical axis is the **audio sending bit rate** (blue line); the right vertical axis is the **audio packet loss rate** (green line).

Different colored dots in the image indicate different types of events, you can filter and view different types of events by clicking the corresponding colored buttons below the image.

8 event types: Entered Room, Connected Signaling Channel Successfully, Establishing Upstream Media Transmission Channel, Open Audio, Established Downstream Media Transmission Channel, Disable Audio, Signal Channel Disconnected, and Exited Voluntarily.

Control Metrics

Same description as Sender - Video Details - Control Metrics.

Audio Capturing Volume

The vertical axis is the audio capturing volume (blue line).
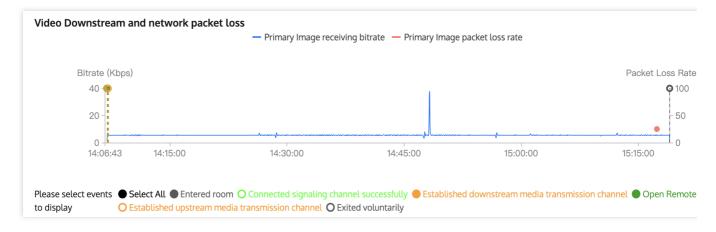

Receiver

Video Details

Video details include 4 major segments: video downsteam and network packet loss, control metrics, video frame rate and lag, and video receive resolution, with multiple metrics in each segment presented as multi-dimensional visual charts.

Video Downsteam and Network Packet Loss

The left vertical axis is the **primary image receiving bit rate** (blue line) and the right vertical axis is the **primary image packet loss rate (**red line).

Different colored dots indicate different types of events, you can filter and view them by clicking the corresponding colored buttons at the bottom of the image.

6 event types: Entered Room, Connected Signaling Channel Successfully, Established Downstream Media Transmission Channel, Open Remote Video, Established Upstream Media Transmission Channel, and Exited Voluntarily.

Video Downstream and network packet loss

— Primary Image receiving bitrate   — Primary Image packet loss rate

Bitrate (Kbps)

Packet Loss Rate

40

100

20

50

0

0

14:06:43   14:15:00   14:30:00   14:45:00   15:00:00   15:15:00

Please select events to display   ● Select All   ● Entered room   ○ Connected signaling channel successfully   ● Established downstream media transmission channel   ● Open Remote   ○ Established upstream media transmission channel   ○ Exited voluntarily

Control Metrics

Same description as Sender - Video Details - Control Metrics.

Video Frame Rate and Lag

The left vertical axis indicates the **primary image frame rate** of received video (blue line); the right vertical axis indicates the **primary image rendering lag duration** (red line); the purple line indicates the **primary image RTC video decoding frame rate**.

VideoFrame Rate and Lag

— Primary ImageRendering Lag Duration   — Primary ImageFrame rate of received video   — Primary ImageWebRTC video decoding frame rate

Frame Rate (fps)

Lag Duration (m

30

20

10

0

14:06:40   14:15:00   14:30:00   14:45:00   15:00:00   15:15:00

Video Receive Resolution

The vertical axis is the primary image received resolution, hover the mouse over the image to view the resolution details.

Audio Details

Audio details include 4 major segments: audio downsteam and network packet loss, control metrics, audio playback volume, and audio playback lag, with multiple metrics in each segment presented as multi-dimensional visual charts.
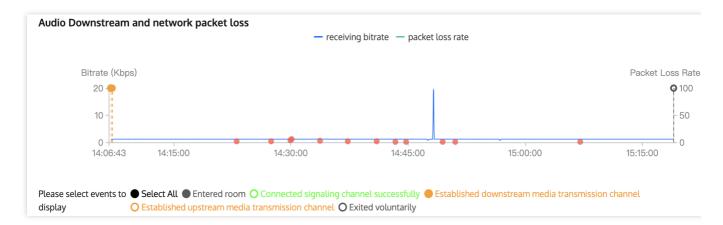
Audio Downsteam and Network Packet Loss

The left vertical axis is the **audio receiving bit rate** (blue line), and the right vertical axis is the **audio packet loss rate** (green line).

Different colored dots in the image indicate different types of events, which can be filtered and viewed by clicking on the corresponding colored buttons at the bottom of the image.

5 event types: Entered Room, Connected Signaling Channel Successfully, Establishing Upstream Media Transmission Channel, Established Downstream Media Transmission Channel, and Exited Voluntarily.

Control Metrics

Same description as Sender - Video Details - Control Metrics.

Audio Playback Volume

The vertical axis is the audio playback volume (blue line).

Audio Playback Lag

Priority Metrics

End-to-end details can be analyzed in terms of **network transmission** and **device status**:

Network Transmission

Ideally, data should be transferred at high bandwidth, with zero packet loss and no delay, but in reality, packet loss, delay, and instability are common, and bandwidth is often limited. Given this, you need to pay special attention to the following points when analyzing network conditions.

Packet loss: In the graph, packet loss is represented by a red line.

Packet Loss Rate	Network Conditions
= 0	Excellent
< 2%	Good
> 5%	Poor
> 10% (or constant packet loss)	Serious network congestion

Bitrate: Normally, the fluctuations of audio/video bitrate should be smaller than 10%. **If the bitrate experiences dramatic fall or fluctuations larger than 30%, it indicates network congestion or jitter**.

Note:

Because the GOP duration of screen sharing is relatively long (5-10 seconds), normally, the bitrate follows a regular cycle, represented by a curve which peaks at keyframes.

Upstream bitrate and packet loss for screen sharing

The bitrate curve of screen sharing follows a cycle with regular peaks.

Frame rate: Normally, video frame rate should stabilize at around 15 fps or higher (5-10 fps for screen sharing). **If the frame rate experiences fluctuations larger than 5 fps or falls and stays below 10 fps, it indicates network congestion or jitter**. When this happens, users experience stutter. Periods of excessively low frame rates are marked red in the graph.

Upstream video frame rate (capture or send)

Video rendering frame rate. Excessively low frame rates are marked red, and the stutter time is provided.

Device Status

Stable device performance is necessary for a successful audio/video call. Preferably, the device uses a small amount of system resources, does not compete for resources with other devices, and collects data without interference. Pay attention to the following aspects when checking device status.

CPU usage: Both system CPU usage and application CPU usage are displayed. Normally, system CPU usage should be lower than 50%. The lower, the better. If **system CPU usage exceeds 85%, the application may stop responding or respond slowly. This is marked by red lines in the graph.**

Time consumption of SDK task: Some Android devices and system versions are unable to calculate CPU usage, in which case you can use **Time Consumption of SDK Task** to assess device performance. If a task takes longer than 60 ms to complete, it indicates high system CPU usage, and the application may not respond or be slow to respond. Consider closing other processes in the background or update your hardware.

Volume:

Capturing volume is the volume of audio captured from the sender's mic. **If it changes, it indicates that the SDK is capturing audio, i.e., the device functions properly.**

Playback volume is the volume of decoded and rendered audio sent to the receiver's speaker. **If it changes, it indicates that the SDK has sent audio to the speaker, i.e., the SDK functions properly.**

The normal volume range is 40-80 dB. If the volume is lower than 40 dB, and the user cannot hear any audio, check for hardware failure or whether the user's phone is muted.

Resolution: The resolutions of video and screen sharing are additional information used mainly to analyze relayed live streaming and the replay of recorded streams. Fluctuations in resolutions indicate that audience watching relayed live streams via CDN or replaying recorded videos, especially web users, may be experiencing player compatibility issues such as stuttering or pixelated video.

Note:

Resolution, bitrate, and frame rate are related to each other. Generally, when resolution is fixed, the higher the bitrate, the clearer the image; when bitrate is fixed, the higher the resolution, the blurrier the image. Set resolution, bitrate, and frame rate properly to ensure good video quality.

Client events: Client events correspond to the calling of SDK APIs by the application and are usually used to help locate software problems, analyze bugs, as well as simulate scenarios by analyzing users' operations. Pay attention to these client events:

Entering/Exiting a room.

Enabling/Disabling the camera or mic.

Device change, such as switching cameras, connecting/disconnecting headphones, and connecting Bluetooth headphones.

Starting/Stopping stream pushing or playback.

Disabling/Enabling audio or video.

Switching networks, for example, from 4G to Wi-Fi.

Click **View Detailed Event** to open the event list and view the operations of key client events.

Real-Time Monitoring

Last updated：2024-09-14 15:32:08

The real-time data monitoring capability offers a real-time, multi-dimensional, and visualized display of the performance of TRTC applications, allowing you to monitor your applications in real time and detect and troubleshoot problems in a timely manner.

Note:

Data monitoring became a paid service on November 1, 2022. The default edition is free. You can purchase the paid edition to unlock more features.

Features

Real-time monitoring:

Statistics including the number of online rooms and users are collected and analyzed automatically at 10-second update intervals.

The real-time monitoring data is linked to the statistics on call details, making it easy for you to troubleshoot user-related issues.

Data is displayed in a visualized way from multiple dimensions. You can view statistics at a specific time point.

Directions

1. Log in to the Tencent RTC Console, select Data Monitoring > Real-Time Monitoring on the left sidebar. If you haven't activated the data monitoring service yet, activate it first.

2. Select an application (`SDKAppID`) to monitor. Please note that you have to wait for about 10 minutes after an application is created before you can view its data.
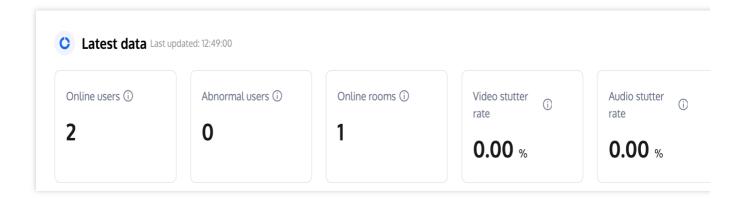
Overview

The data on the real-time monitoring page is updated every 20 seconds (you can find the last updated time next to **Latest data**).

Select application: Select an application to view the data of all rooms under the application.

Pause data: Click **Pause data** to lock data at the current time, and click **View latest data** to unlock it.

Latest data

The **Latest data** section shows online users, abnormal users, online rooms, video stutter rate, and audio stutter rate in the last 20 seconds. For details, see monitoring indicators. The data is updated every 20 seconds.
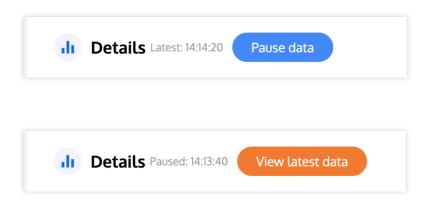

Details

The **Details** section shows data for five different scale- and quality-related indicators. You can pause the data at the current time.

time state

Users can switch to the paused state or the latest state by clicking the button **Pause data** / **View latest data**, or switch to the paused state by double clicking the node in the image. In the locked state, the data will be fixed at a

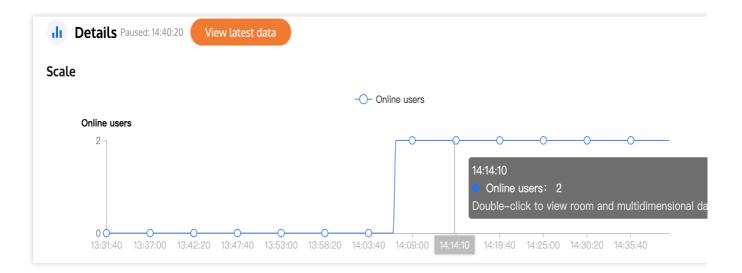specific moment; in the latest state, it's updated every 20 seconds.

Scale indicators

This area uses line charts to show the number of online users (abnormal users) and rooms of the selected application in the last 64 minutes. The data is updated every 10 seconds. You can hover over anywhere on the line charts to view data at a specific time point and double-click a chart to pause data at the current time.

Quality indicators

This area uses line charts to show the average video and audio stutter rate and number of abnormal users in the last 64 minutes across all rooms of the selected application. The data is updated every 10 seconds. You can hover over anywhere on the line charts to view data at a specific time point and double-click a chart to pause data at the current time.

Example: The following image shows part of the page in a paused state.

Room data

This area shows the Room ID, Online users, Abnormal users, Video stutter rate, Audio stutter rate, and Operation of active rooms.

Pause data: Support pause data to observe the room details at a specific time point, users can click the button **Pause data** to switch to the pause state, and click again to return to the latest state.

Download room data: Click **Download room data** to download the xlsx format file of room data at the current time.

Abnormal room data: Click **Exception** option under Operation field, user can view the abnormal room data, including 8 fields: Room ID, User ID, Exception, Region, Network, Device, SDK Version, ISP. user can filter the 8 dimensions to search for the abnormal data.

Call details: Click **Call Details** option under Operation field to view the call quality data of users in a room. See Call details for more description.

In the actions above, Download room data, Abnormal room data and Call details will pause data at the current time. To resume viewing the latest data, click **View latest data** next to **Details**.

Multidimensional data

This area shows data from different dimensions such as region, network, device, SDK version, and ISP.

Support pause data to observe information at a specific point in time.

You can also download the data from this area.

Key Indicators

Scale indicators

Indicator	Description
Online users	The total number of users who entered the rooms of the selected application in the past 10 seconds. A user who entered multiple rooms is counted multiple times. A user who entered the same room multiple times is counted only once.

Online rooms	The total number of rooms that were entered by users in the past 10 seconds
Abnormal users	The total number of abnormal users (counted by user IDs) in the rooms of the selected application in the past 10 seconds. Abnormal users are users who encountered an exception. Exceptions: no audio, abnormal video frame rate, high CPU usage.

Quality indicators

Indicator	Description
Video stutter rate	Stuttering duration (any stutter that lasts for 600 ms or longer)/Total video duration x 100%, updated every 10 seconds
Audio stutter rate	Stuttering duration (any stutter that lasts for 200 ms or longer is counted)/Total audio duration x 100%, updated every 10 seconds
Video stutter severity	Low: [0.00% - 5.00%) (green). Moderately high: [5.00%, 10.00%) (yellow). High: ≥ 10.00% (red).
Audio stutter severity	Low: [0.00% - 3.00%) (green). Moderately high: [3.00% - 5.00%) (yellow). High: ≥ 5.00% (red).

RESTful APIs

You can also use the RESTful APIs we provide to query real-time monitoring data.

Note:

Only users who have activated the **Standard** edition of the dashboard can use the RESTful APIs.

Data Dashboard

Last updated：2025-02-25 16:47:49

The **Data Insights** on RTC Analytics provides features for viewing and analyzing the scale and quality data of business operation dashboard from an application dimension. It helps you quickly understand the call scale data and trends under the application (sdkappid) while offering insights into call quality across multiple dimensions.

Note:

The Data Dashboard is a paid feature and requires a subscription to the paid version to unlock additional features. For more information, see Version Features and Billing Information of Data Monitoring.

Feature Overview

Features of the Data Dashboard:

Dashboard Data: Supports viewing the number of users, the number of rooms, the number of users during peak hours, and the number of rooms during peak hours over a specified period, providing a quick overview of the operational scale.

Multi-dimensional Quality Analysis: Supports viewing room call quality across various dimensions over a specified period, including metrics such as video stutter rate and audio stutter rate, enabling timely insights into call quality across different dimensions in business operations.

Directions

1. Log in to the Tencent RTC Console > RTC Analytics > Data Insights, go to Data Monitoring and activate the service. If it is already activated, you can skip this step.

Note:

To use this feature, activate the Data Monitoring service in advance. The data query range for the Data Dashboard starts from the next day after the service activation. Historical data prior to the service activation is not included in the statistics and cannot be retrieved.

2. Go to RTC Analytics > Data Insights to access the Data Dashboard page.

3. Filter the desired application (sdkappid) and select the time range to view. **Currently, the free version (default) allows filtering for up to the past 7 days**, with a maximum range of 60 days available by unlocking the RTC Analytics Monthly Packages.

Feature Description

Dashboard Metrics

Statistical Metrics: The dashboard metrics provide four key operational scale metrics: the number of users, the number of rooms, the number of users during peak hours, and the number of rooms during peak hours. These metrics help you quickly understand the operational scale of the current application within the selected time range.

Metric	Description
The number of	The total number of users who joined a room during the selected time range (deduplicated

| users	by userID and the combination of create_time and roomID).
The number of rooms	The total number of rooms created during the selected time range (deduplicated by the combination of create_time and roomID).
The number of users during peak hours	The maximum number of users in a room simultaneously during the selected time range, calculated at a 1-minute granularity.
The number of rooms during peak hours	The maximum number of rooms existing simultaneously during the selected time range, calculated at a 1-minute granularity.
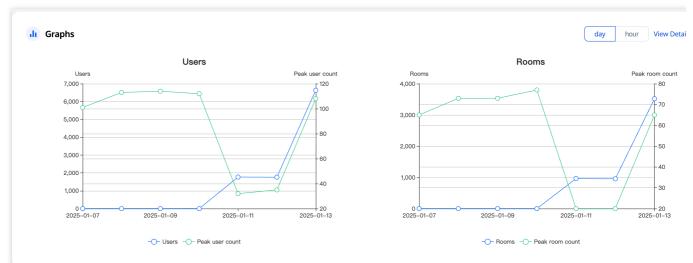
Note:

For the number of users, if the same user (userID) joins different roomID, it will be counted as two users; similarly, if the same user (userID) joins the same roomID created at different times, it will also be counted as two users.

Graphs: They provide trend line charts for the above metrics, supporting both daily and hourly views:

If you view by **day**, you can see the scale metrics for that day at each coordinate point.

If you view by **hour**, only hourly granularity data for a single day is supported, with Indicator Statistical Rules as described above. When you switch to **hourly** granularity, the display automatically switches to the most recent day within your selected date range.

Note:

Since there may be call rooms spanning across hours, the number of users in the same room may be counted separately in two adjacent hours. As a result, summing up hourly granularity data for a day may yield a higher total number of users and rooms than the daily dimension statistics.

You can click **View Details** to access the corresponding data details, with an option to export the data.

Details   ✕

Download

Peak room count ⓘ

77

Date	Users	Rooms	Peak user count	Peak room count
Jan 07, 2025 (UTC+08:00)	0	0	101	65
Jan 08, 2025 (UTC+08:00)	0	0	113	73
Jan 09, 2025 (UTC+08:00)	0	0	114	73
Jan 10, 2025 (UTC+08:00)	0	0	112	77
Jan 11, 2025 (UTC+08:00)	1770	963	32	20
Jan 12, 2025 (UTC+08:00)	1763	954	35	20
Jan 13, 2025 (UTC+08:00)	6634	3519	108	65

day   hour   View Detai

Peak room count

Total items: 7   10 ▼ / page |◄ ◄   1   / 1 page   ► ►|

Confirm   Cancel

Dimensional Data

Quality: The quality under dimensional data provides the feature to query call quality metrics across various dimensions (such as region, SDK version, and network type). You can view the desired quality metrics under the corresponding dimension module and click **View Details** to access all detailed data, as shown below.

You can click **View Details** to filter by the corresponding dimensions and access detailed data.

Quality Metrics

Metric	Description

Video stutter rate	The duration of video stutters / the total video duration x 100%. A video stutter lasting more than 600 ms is included in the stutter duration.
Audio stutter rate	The duration of audio stutters / the total audio duration x 100%. An audio stutter lasting more than 200 ms is included in the stutter duration.
Lag rate	The duration of end-to-end network lag / the total audio and video duration x 100%. An end-to-end network lag exceeding 400 ms is included in the lag duration.
Successful room entries	The number of users who successfully joined a room / the total number of users who attempted to join x 100%.
Room entries within 5s	The number of users who successfully joined a room within 5 seconds / the total number of users who attempted to join x 100%.

Dimensions

Dimension	Description
Distribution by region	The province where the user is located, distinguishing between regions of the Chinese mainland and regions outside the Chinese mainland.
Device type	The type of device used by the user.
SDK Version	The SDK version used by the user during the call.
Operator	The network operator used by the user.
Network type	The type of network used during the call, such as wired, Wi-Fi, and 4G.

Restful API

The Data Dashboard feature provides RESTful APIs, allowing you to retrieve relevant data through API requests.
Note:

The Data Dashboard RESTful API is only available to customers who have subscribed to the **Basic Edition** or **Advanced Edition** of the Data Monitoring service. For details on enabling the Data Monitoring service, see Version Features and Billing Instructions.

Query TRTC Data Dashboard Scale Metrics
Query TRTC Data Dashboard Quality Metrics

Development Assistance
UserSig Generation and Verification

Last updated：2025-01-14 09:46:23

You can generate UserSig online in the TRTC console, but it should be used only for quick testing at the development stage. This avoids key leakage and prevents attackers from stealing your traffic.

Signature (UserSig) Generator

Signatures (UserSig) allow you to build trust with Tencent Cloud.

1. Log in to the Tencent RTC console, and select **Development Tools > UserSig Tools** on the left sidebar.

2. In the **Signature (UserSig) Generator** module on the left, select the application (SDKAppID) you created from the drop-down list. A secret key (`key`) is generated automatically.

3. Enter the user name (`UserID`).

4. Click **Generate** to generate your UserSig.

Signature (UserSig) Verifier

This is used to check the validity of your signature (UserSig).

Note:

Make sure that you enter the correct SDKAppID and UserID for the UserSig you want to verify.

1. Log in to the TRTC console, select **Usersig Tools** on the left sidebar.

2. In the **Signature (UserSig) Verifier** module on the right, select the application (`SDKAppID`) whose signature you want to verify. A secret key is generated automatically.

3. Enter the user name (`UserID`).

4. Copy and paste the signature (UserSig) that needs verification to **Signature (UserSig)**, and click **Verify Now.**

Note:

If your UserSig is generated in **Signature (UserSig) Generator**, click **Copy Signature (UserSig)** to copy the signature.

5. View the verification results.

Verify result

Verify result：Verified successfully
Generate ticket parameters：
    SDKAppid：
    Identifier：
    Creation time：2023-12-11 19:52:14
    Validity：180day 0hr 0min 0sec
    Expiration time：2024-06-08 19:52:14

RTMP Address Generator

Last updated：2025-01-17 15:25:51

The TRTC console supports generating streaming addresses for the RTMP streaming room entry feature, which is only suitable for the testing phase. For official online business, please deploy the address generation logic to backend services to avoid traffic theft caused by encryption key leak.

Generating Stream Address

1. Go to the Tencent RTC console, select **Development Tools > RTMP Address Generator** in the left sidebar, and check the **Generating Stream Address** module.
2. Click the dropdown box to select your created application (SDKAppID).
3. Fill in **Push Roomname (RoomID)** and **Push Username (UserID)**.
4. Click **Generate** to immediately generate the corresponding RTMP streaming address.

← RTMP Address Generator

ⓘ This page is used to generate the Push address for **RTMP Streaming with TRTC** ⧉ function, which is applicable to the testing phase. For official online business, please deploy the address generation logic to the backend service to calculate and generate, in order to avoid the leak of encryption key leading to traffic theft.

🅰 Push address

Application (SDKAppID)	[blurred] ▾
Push Roomname (RoomID)	test ⓘ
Push Username (UserID)	user ⓘ

Generate   Splicing Address Explanation

Generate result

RTMP Streaming with TRTC ⧉

RTMP Push Address	[blurred]	⎘ Copy
OBS Server	[blurred]	⎘ Copy
OBS Streaming Secret Key	[blurred]	⎘ Copy

Enter the room authorization parameter Usersig Validity default is 180 days.