

# **Tencent Real-Time Communication**

## **Console Guide**

### **Product Documentation**



## Copyright Notice

©2013-2022 Tencent Cloud. All rights reserved.

Copyright in this document is exclusively owned by Tencent Cloud. You must not reproduce, modify, copy or distribute in any way, in whole or in part, the contents of this document without Tencent Cloud's the prior written consent.

## Trademark Notice



All trademarks associated with Tencent Cloud and its services are owned by Tencent Cloud Computing (Beijing) Company Limited and its affiliated companies. Trademarks of third parties referred to in this document are owned by their respective proprietors.

## Service Statement

This document is intended to provide users with general information about Tencent Cloud's products and services only and does not form part of Tencent Cloud's terms and conditions. Tencent Cloud's products or services are subject to change. Specific products and services and the standards applicable to them are exclusively provided for in Tencent Cloud's applicable terms and conditions.

# Contents

## Console Guide

### Monitoring Dashboard

#### Call Data

##### Call List

##### Call Details

##### End-to-End Details

### UserSig Generation and Verification

# Console Guide

## Monitoring Dashboard

### Call Data

### Call List

Last updated : 2022-09-26 16:32:53

TRTC 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 TRTC console, and click [Monitoring Dashboard](#) to view the call details of all rooms under the logged-in account. By default, the call list displays the calls made under the last created application during the day of query. The most recent calls appear at the top of the list.

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
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 <a href="#">View Call Details</a> 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 **Last 14 days**, 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**.

## More Operations

If you want to view the call details of a particular room, see [Call Details](#).

# Call Details

Last updated : 2022-09-26 16:33:30

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 TRTC 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 <b>Add user</b> or <b>View All Users</b> . 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 platform or OS used by the user
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 select a specific remote user to view the data received from that 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.

# End-to-End Details

Last updated : 2022-09-26 16:35:48

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 TRTC 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.

You can analyze end-to-end data from [network conditions](#) and [device status](#).

### Analyzing network conditions

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.**

### notice

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.

## View 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.

### explain

[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.

# UserSig Generation and Verification

Last updated : 2022-10-28 15:38:23

You can generate UserSig online in the TRTC console, but it should be used only for quick testing at the development stage. For official launch, please [calculate UserSig on the server](#). 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 TRTC console, select **Application Management** on the left sidebar, and click UserSig generation.
2. In **Signature (UserSig) Generator**, 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 Signature (UserSig)** to generate your UserSig.

## Signature (UserSig) Checker

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

### notice

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 **Application Management** on the left sidebar, and click UserSig verification.
2. In **Signature (UserSig) Checker**, 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**.

### explain

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

5. View the verification results.

## Reference

For more information about UserSig, see [FAQs \(UserSig\)](#).