Tencent Real-Time Communication

Advanced Features

Product Documentation

Contents

Advanced Features
Relay to CDN

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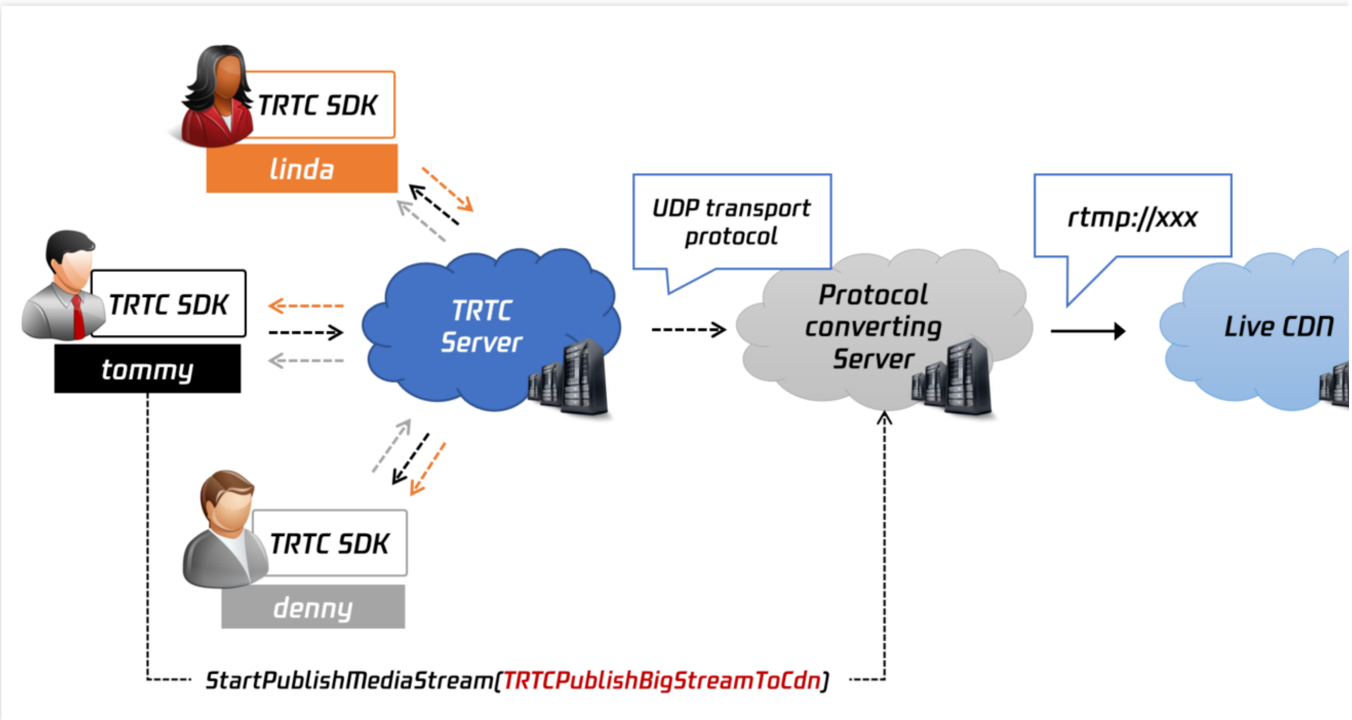
This document describes how to publish (relay) audio/video streams in TRTC to CDNs so that viewers can watch the streams using standard live streaming players.

Publishing the Local User's Stream to CDNs



Prerequisites

1. Go to activate the Tencent Cloud CSS service (please note that your account has been bound to the card and and has no outstanding charges, otherwise, the CSS service cannot be activated). Configuring a playback domain is mandatory for CSS. For detailed instructions, please refer to Adding Your Own Domain.
2. Select **Domain Management** in the left navigation bar, and you will see a new push domain name added to your domain list, formatted as `xxxxx.livepush.myqcloud.com` , where xxxxx represents a number called bizid.

3. Click **Add Domain**, enter the playback domain name you have registered, select the domain name type as **Playback Domain**, select the acceleration region, and click **Confirm**.
4. After the domain name is added successfully, the system will automatically assign you a CNAME domain name (ending with `.liveplay.myqcloud.com`). The CNAME domain name cannot be accessed directly. You need to complete the CNAME configuration with your domain name service provider. After the configuration takes effect, you can enjoy the CSS service. For detailed instructions, please refer to CNAME Configuration.

Note：

You do not need to add a push domain name. After enabling the relayed live streaming feature in Step 1, Tencent Cloud will add a push domain name formatted as `xxxxx.livepush.myqcloud.com` to your CSS console by default. This domain name serves as a default push domain name agreed upon between CSS service and TRTC service.

Description

You can use the **startPublishMediaStream** API of `TRTCCloud` to publish the audio/video streams of local users to live streaming CDNs (known in TRTC as "relay to CDN").

The TRTC server will send the audio/video data directly to the CDN server. Because the data is not transcoded, the cost is relatively low.

However, if there are multiple users publishing audio/video streams in a room, there will be a CDN stream for each user. Multiple players are needed to play the streams, and they may not play in sync.

Directions

Follow the steps below to publish the local user’s stream to CDNs.

1. Create a `TRTCPublishTarget` object and set `mode` in the object to `TRTCPublishBigStreamToCdn` or `TRTCPublishSubStreamToCdn`. The former is used to publish the user’s primary stream (usually the camera), and the latter is used to publish the user’s substream (usually the screen).
2. Set `cdnUrlList` in the `TRTCPublishTarget` object to one or multiple CDN addresses (which usually starts with `rtmp://`). If you publish to the Tencent Cloud CDN, set `isInternalLine` to `true`; otherwise, set it to `false`.
3. Because the data is not transcoded, leave `TRTCStreamEncoderParam` and `TRTCStreamMixingConfig` empty.
4. Call `startPublishMediaStream`. If the `taskId` parameter returned by the `onStartPublishMediaStream` callback is not empty, the local API call is successful.
5. To stop publishing, call `stopPublishMediaStream`, passing in the `taskId` returned by `onStartPublishMediaStream`.

Sample code

The code below publishes the local user’s stream to a live streaming CDN.

Java
ObjC
C++
Dart

```
TRTCCloudDef.TRTCPublishTarget target = new TRTCCloudDef.TRTCPublishTarget();
target.mode = TRTC_PublishBigStream_ToCdn;

TRTCCloudDef.TRTCPublishCdnUrl cdnUrl= new TRTCCloudDef.TRTCPublishCdnUrl();
cdnUrl.rtmpUrl = "rtmp://tencent/live/bestnews";
cdnUrl.isInternalLine = true;
```

```
target.cdnUrlList.add(cdnUrl);

mTRTCCloud.startPublishMediaStream(target, null, null);
```

```
TRTCPublishTarget* target = [[TRTCPublishTarget alloc] init];
target.mode = TRTCPublishBigStreamToCdn;

TRTCPublishCdnUrl* cdnUrl = [[TRTCPublishCdnUrl alloc] init];
cdnUrl.rtmpUrl = @"rtmp://tencent/live/bestnews";
cdnUrl.isInternalLine = YES;
```



```
NSMutableArray* cdnUrlList = [NSMutableArray new];
[cdnUrlList addObject:cdnUrl];
target.cdnUrlList = cdnUrlList;

[_trtcCloud startPublishMediaStream:target encoderParam:nil mixingConfig:nil];
```

```
TRTCPublishTarget target;
target.mode = TRTCPublishMode::TRTCPublishBigStreamToCdn;
```

```
TRTCPublishCdnUrl* cdn_url_list = new TRTCPublishCdnUrl[1];
cdn_url_list[0].rtmpUrl = "rtmp://tencent/live/bestnews";
cdn_url_list[0].isInternalLine = true;

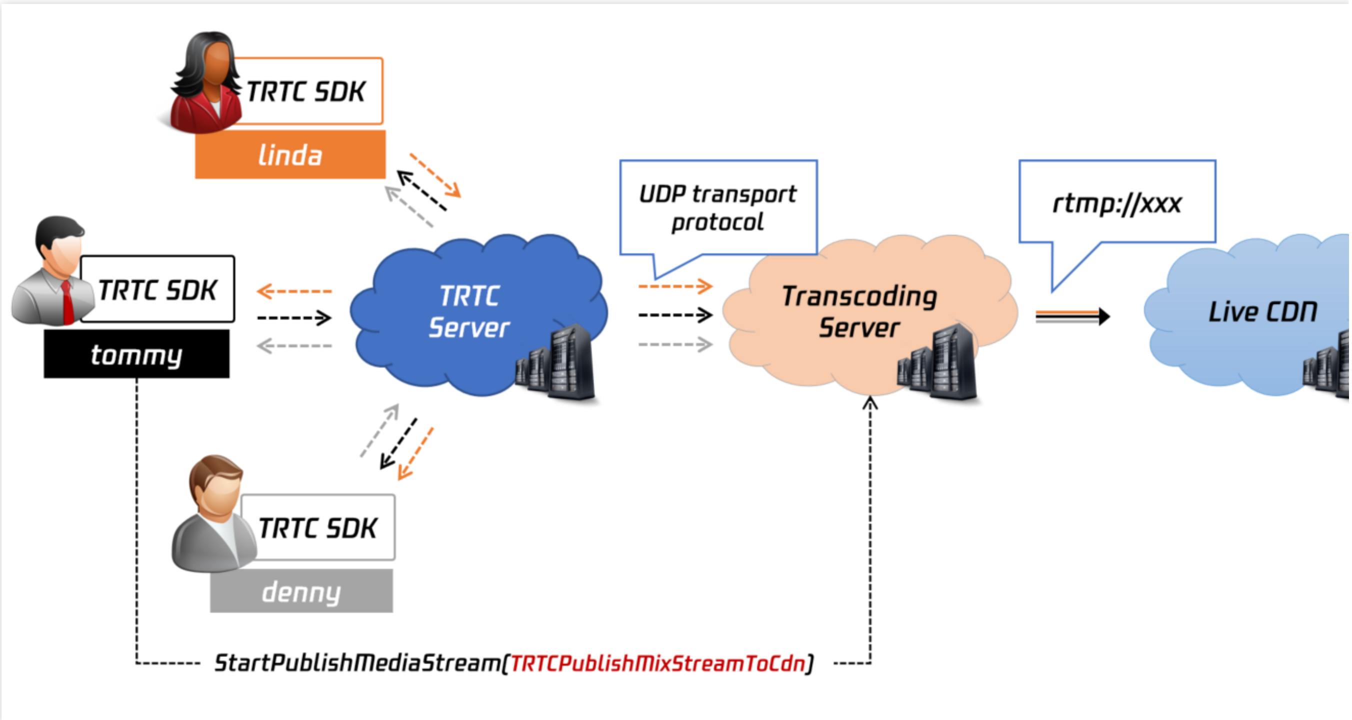
target.cdnUrlList = cdn_url_list;
target.cdnUrlListSize = 1;

trtc->startPublishMediaStream(&target, nullptr, nullptr);
delete[] cdn_url_list;
```

```
TRTCPublishTarget target = TRTCPublishTarget();
target.mode = TRTCPublishMode.TRTCPublishBigStreamToCdn;
TRTCPublishCdnUrl cdnUrlEntity = new TRTCPublishCdnUrl();
cdnUrlEntity.rtmpUrl = "rtmp://tencent/live/bestnews";
cdnUrlEntity.isInternalLine = true;
target.cdnUrlList.add(cdnUrlEntity);

trtcCloud.startPublishMediaStream(target: target);
```

Publishing Mixed Streams to CDNs



Description

You can call **startPublishMediaStream** to mix the streams of multiple users in a TRTC room into one stream and publish the stream to a CDN. The `TRTCStreamEncoderParam` and `TRTCStreamMixingConfig` parameters of the API allow you to determine the details of stream mixing and transcoding.
The streams will be decoded on the cloud first, mixed, and then re-encoded according to the stream mixing parameters (`TRTCStreamMixingConfig`) and transcoding parameters (`TRTCStreamEncoderParam`) you specify. Afterward, they will be published to CDNs. In this mode, additional transcoding fees are charged.

Directions

Follow the steps below to mix the streams of multiple users in a room and publish the mixed stream to CDNs.

1. Create a `TRTCPublishTarget` object and set `mode` in the object to `TRTCPublishMixStreamToCdn` .
2. Set `cdnUrlList` in the `TRTCPublishTarget` object to one or multiple CDN addresses (which usually start with `rtmp://`). If you publish to the Tencent Cloud CDN, set `isInternalLine` to `true` ; otherwise, set it to `false` .
3. Set the encoding parameters (`TRTCStreamEncoderParam`):

Video encoding parameters: Specify the resolution, frame rate (15 fps is recommended), bitrate, and GOP (3 seconds is recommended). Bitrate and resolution work in correlation with each other. The table below lists some recommended resolution and bitrate settings.

Audio encoding parameters: Specify the codec, bitrate, sample rate, and sound channels according to the `AudioQuality` value you pass in when calling `startLocalAudio` .

videoEncodedWidth	videoEncodedHeight	videoEncodedFPS	videoEncodedGOP	videoEncodedKbp
640	360	15	3	800 Kbps
960	540	15	3	1200 Kbps
1280	720	15	3	1500 Kbps
1920	1080	15	3	2500 Kbps

TRTCAudioQuality	audioEncodedSampleRate	audioEncodedChannelNum	audioEncodedKbps
TRTCAudioQualitySpeech	48000	1	50
TRTCAudioQualityDefault	48000	1	50
TRTCAudioQualityMusic	48000	2	60

4. Set the parameters for audio mixing and video layout (TRTCStreamMixingConfig):

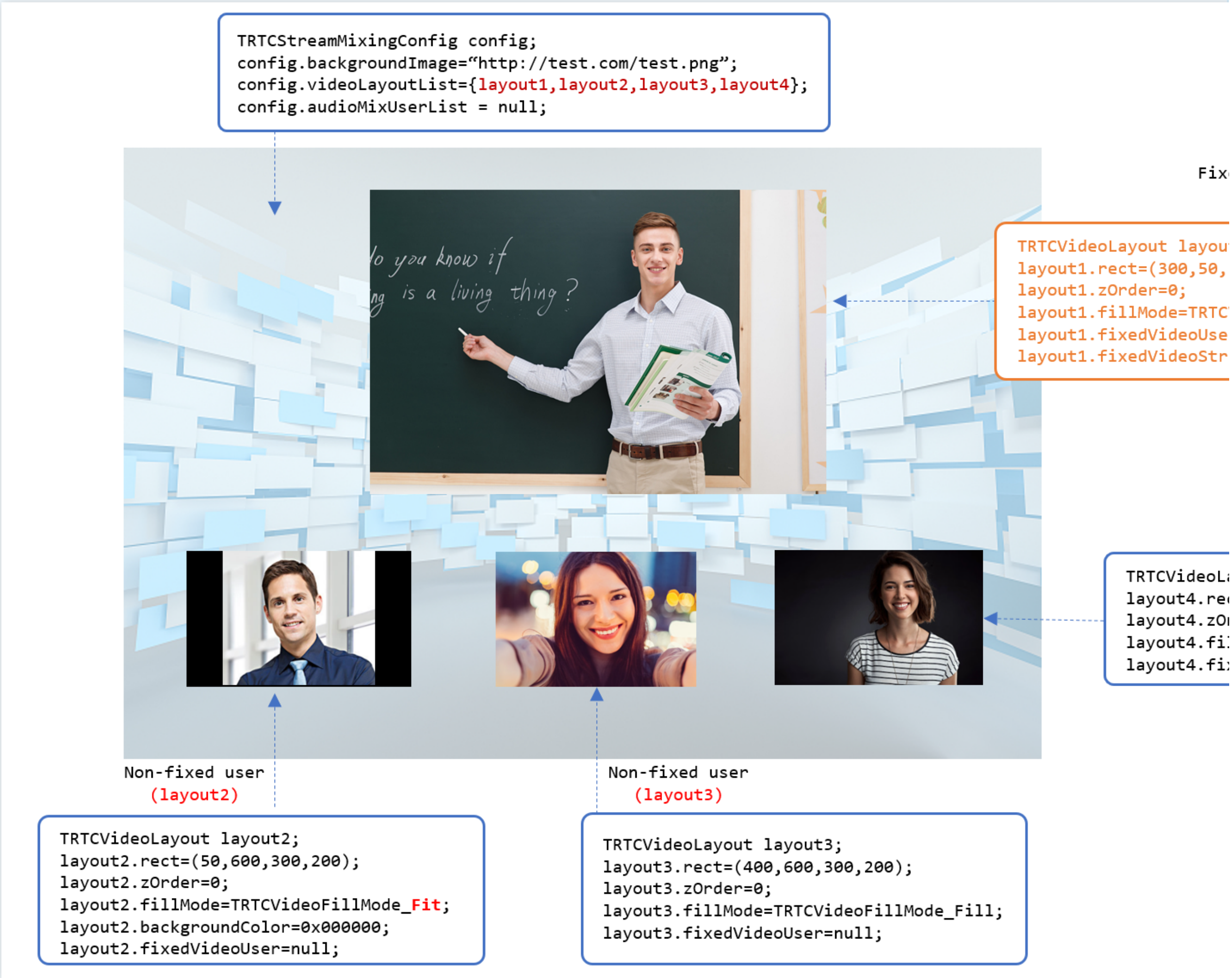
Audio mixing parameters (audioMixUserList): You can leave this parameter empty to mix all audios in a room, or you can set it to the IDs of users whose audios you want to mix.

Video layout parameters (videoLayoutList): Video layout is determined by an array. Each TRTCVideoLayout element in the array determines the position, dimensions, and background color of a video window. If you specify fixedVideoUser, the window defined by the TRTCVideoLayout element will display the video of a specific user. If you set fixedVideoUser to null, the TRTC server will determine whose video to display in the window.

Example 1: Mix four users' streams and use an image as the background.

`layout1` specifies the position (upper half of the canvas) and dimensions (640 x 480) of the camera video of user `jerry` .

Because no user IDs are specified for `layout2` , `layout3` , and `layout4` , TRTC will display the videos of the other three users in the windows based on its own rule.

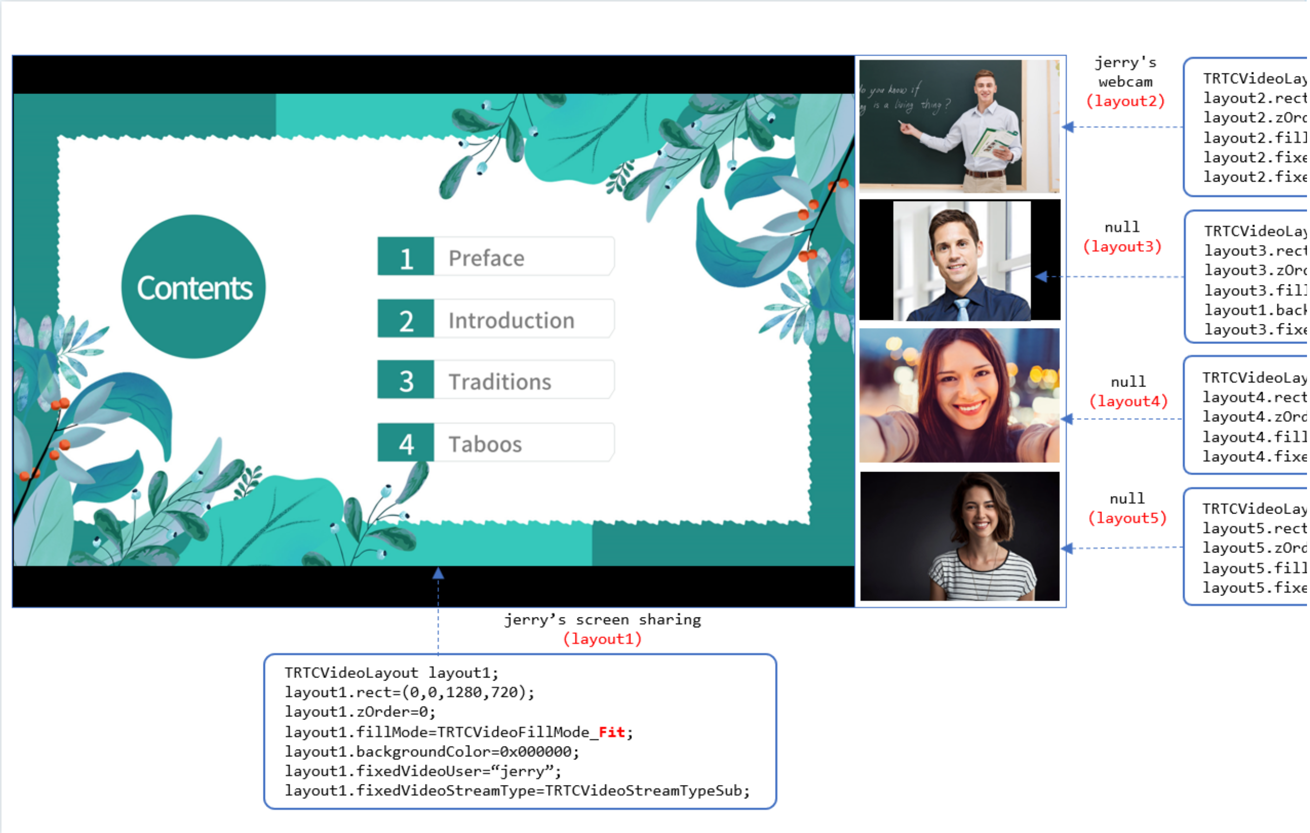


Example 2: Mix the camera video and screen of one user plus the camera videos of three other users.

`layout1` specifies the position (left) and dimensions (1280 x 720) of user `jerry` 's screen. The rendering mode used is aspect fit (`Fit`), and the background color is black.

`layout2` specifies the position (top right) and dimensions (300 x 200) of user `jerry` 's camera video. The rendering mode used is aspect fill (`Fill`).

Because no user IDs are specified for `layout3` , `layout4` , and `layout5` , TRTC will display the videos of the other three users in the windows based on its own rule.



5. Call startPublishMediaStream. If the taskId parameter returned by the onStartPublishMediaStream callback is not empty, the local API call is successful.

6. To change the stream mixing parameters (for example, the video layout), call `updatePublishMediaStream`, passing in the `taskId` returned in step 6 as well as the new `TRTCStreamMixingConfig` parameters. We recommend you do not change `TRTCStreamEncoderParam` during relay because doing so will affect the stability of CDN playback.

7. To stop publishing, call `stopPublishMediaStream`, passing in the `taskId` returned by `onStartPublishMediaStream`.

Sample code

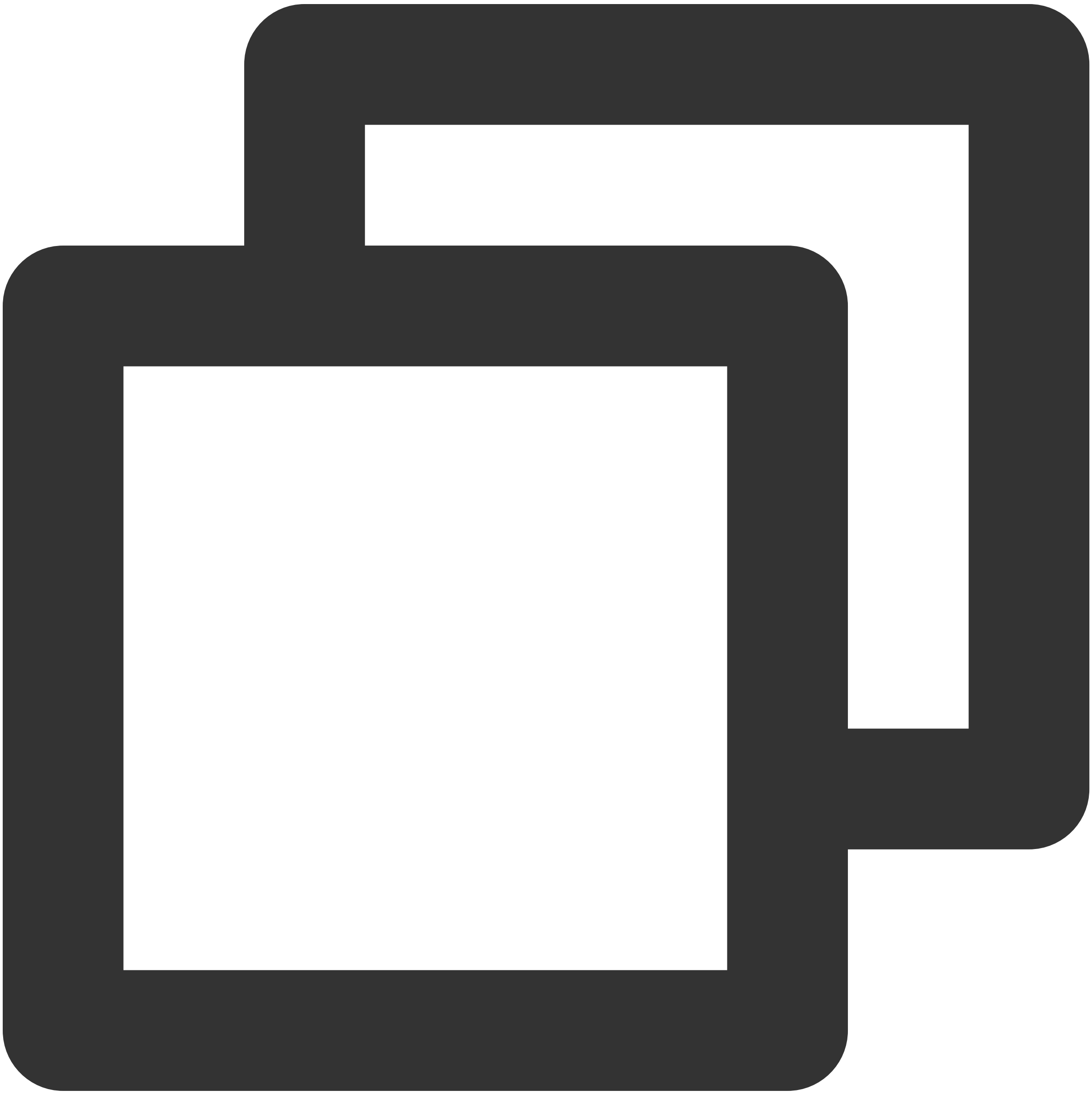
The code below mixes the streams of multiple users in a room and publishes the result to a CDN.

Java

ObjC

C++

Dart



```
TRTCCloudDef.TRTCPublishTarget target = new TRTCCloudDef.TRTCPublishTarget();
target.mode = TRTC_PublishMixedStream_ToCdn;

TRTCCloudDef.TRTCPublishCdnUrl cdnUrl= new TRTCCloudDef.TRTCPublishCdnUrl();
cdnUrl.rtmpUrl = "rtmp://tencent/live/bestnews";
cdnUrl.isInternalLine = true;
target.cdnUrlList.add(cdnUrl);


TRTCCloudDef.TRTCStreamEncoderParam encoderParam
    = new TRTCCloudDef.TRTCStreamEncoderParam();
```

```
encoderParam.videoEncodedWidth = 1280;
encoderParam.videoEncodedHeight = 720;
encoderParam.videoEncodedFPS = 15;
encoderParam.videoEncodedGOP = 3;
encoderParam.videoEncodedKbps = 1000;
encoderParam.audioEncodedSampleRate = 48000;
encoderParam.audioEncodedChannelNum = 1;
encoderParam.audioEncodedKbps = 50;
encoderParam.audioEncodedCodecType = 0;

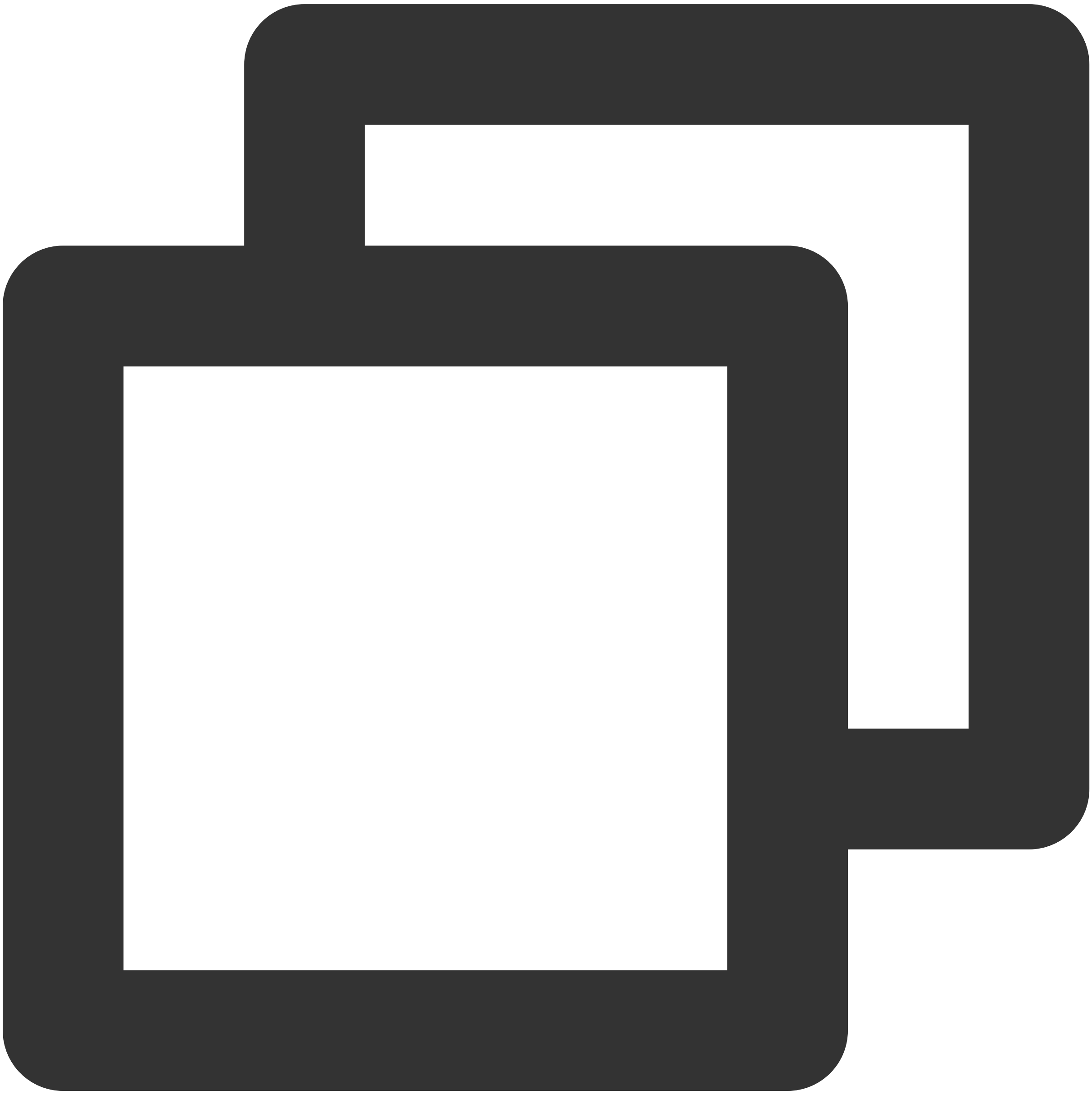
TRTCCloudDef.TRTCStreamMixingConfig mixingConfig =
      new TRTCCloudDef.TRTCStreamMixingConfig();
TRTCCloudDef.TRTCVideoLayout layout1 = new TRTCCloudDef.TRTCVideoLayout();
layout1.zOrder = 0;
layout1.x = 0;
layout1.y = 0;
layout1.width = 720;
layout1.height = 1280;
layout1.fixedVideoStreamType = TRTCCloudDef.TRTC_VIDEO_STREAM_TYPE_SUB;
layout1.fixedVideoUser.intRoomId = 1234;
layout1.fixedVideoUser.userId = "mike";

TRTCCloudDef.TRTCVideoLayout layout2 = new TRTCCloudDef.TRTCVideoLayout();
layout2.zOrder = 0;
layout2.x = 1300;
layout2.y = 0;
layout2.width = 300;
layout2.height = 200;
layout2.fixedVideoStreamType = TRTCCloudDef.TRTC_VIDEO_STREAM_TYPE_BIG;
layout2.fixedVideoUser.intRoomId = 1234;
layout2.fixedVideoUser.userId = "mike";

TRTCCloudDef.TRTCVideoLayout layout3 = new TRTCCloudDef.TRTCVideoLayout();
layout3.zOrder = 0;
layout3.x = 1300;
layout3.y = 220;
layout3.width = 300;
layout3.height = 200;
layout3.fixedVideoStreamType = TRTCCloudDef.TRTC_VIDEO_STREAM_TYPE_SUB;
layout3.fixedVideoUser = null;

mixingConfig.videoLayoutList.add(layout1);
mixingConfig.videoLayoutList.add(layout2);
mixingConfig.videoLayoutList.add(layout3);
mixingConfig.audioMixUserList = null;

mTRTCCloud.startPublishMediaStream(target, encoderParam, mixingConfig);
```



```
TRTCPublishTarget* target = [[TRTCPublishTarget alloc] init];
target.mode = TRTCPublishMixStreamToCdn;

TRTCPublishCdnUrl* cdnUrl = [[TRTCPublishCdnUrl alloc] init];
cdnUrl.rtmpUrl = @"rtmp://tencent/live/bestnews";
cdnUrl.isInternalLine = YES;

NSMutableArray* cdnUrlList = [NSMutableArray new];
[cdnUrlList addObject:cdnUrl];
```

```
target.cdnUrlList = cdnUrlList;

TRTCStreamEncoderParam* encoderParam = [[TRTCStreamEncoderParam alloc] init];
encoderParam.videoEncodedWidth = 1280;
encoderParam.videoEncodedHeight = 720;
encoderParam.videoEncodedFPS = 15;
encoderParam.videoEncodedGOP = 3;
encoderParam.videoEncodedKbps = 1000;
encoderParam.audioEncodedSampleRate = 48000;
encoderParam.audioEncodedChannelNum = 1;
encoderParam.audioEncodedKbps = 50;
encoderParam.audioEncodedCodecType = 0;

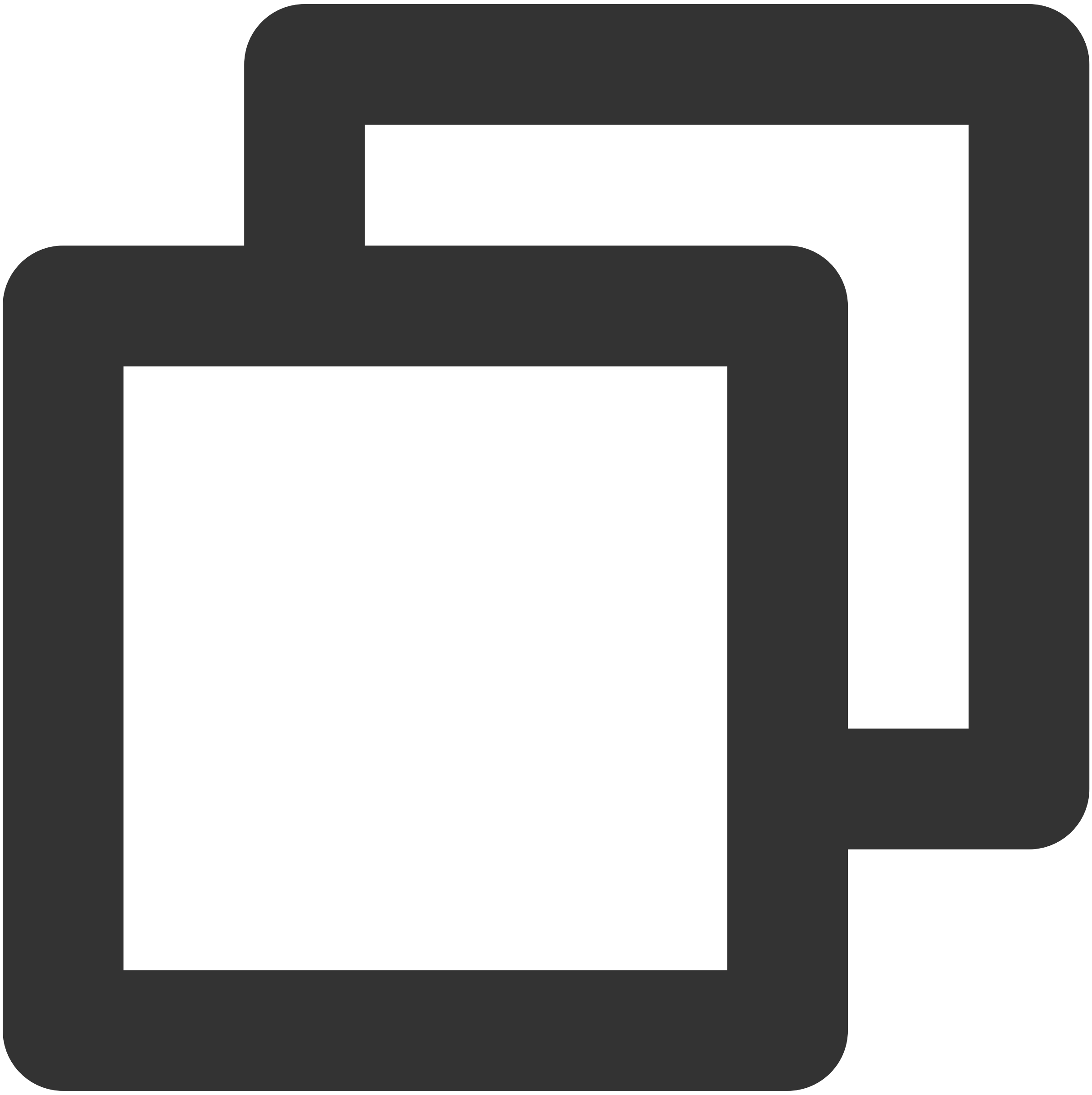
TRTCStreamMixingConfig* config = [[TRTCStreamMixingConfig alloc] init];
NSMutableArray* videoLayoutList = [NSMutableArray new];
TRTCVideoLayout* layout1 = [[TRTCVideoLayout alloc] init];
layout1.zOrder = 0;
layout1.rect = CGRectMake(0, 0, 720, 1280);
layout1.fixedVideoStreamType = TRTCVideoStreamTypeSub;
layout1.fixedVideoUser.intRoomId = 1234;
layout1.fixedVideoUser.userId = @"mike";

TRTCVideoLayout* layout2 = [[TRTCVideoLayout alloc] init];
layout2.zOrder = 0;
layout2.rect = CGRectMake(1300, 0, 300, 200);
layout2.fixedVideoStreamType = TRTCVideoStreamTypeBig;
layout2.fixedVideoUser.intRoomId = 1234;
layout2.fixedVideoUser.userId = @"mike";

TRTCVideoLayout* layout3 = [[TRTCVideoLayout alloc] init];
layout3.zOrder = 0;
layout3.rect = CGRectMake(1300, 220, 300, 200);
layout3.fixedVideoStreamType = TRTCVideoStreamTypeSub;
layout3.fixedVideoUser = nil;

[videoLayoutList addObject:layout1];
[videoLayoutList addObject:layout2];
[videoLayoutList addObject:layout3];
config.videoLayoutList = videoLayoutList;
config.audioMixUserList = nil;

[_trtcCloud startPublishMediaStream:target encoderParam:encoderParam mixingConfig:c
```



```
TRTCPublishTarget target;
target.mode = TRTCPublishMode::TRTCPublishMixStreamToCdn;

TRTCPublishCdnUrl* cdn_url = new TRTCPublishCdnUrl[1];
cdn_url[0].rtmpUrl = "rtmp://tencent/live/bestnews";
cdn_url[0].isInternalLine = true;
target.cdnUrlList = cdn_url;
target.cdnUrlListSize = 1;

TRTCStreamEncoderParam encoder_param;
```

```
encoder_param.videoEncodedWidth = 1280;
encoder_param.videoEncodedHeight = 720;
encoder_param.videoEncodedFPS = 15;
encoder_param.videoEncodedGOP = 3;
encoder_param.videoEncodedKbps = 1000;
encoder_param.audioEncodedSampleRate = 48000;
encoder_param.audioEncodedChannelNum = 1;
encoder_param.audioEncodedKbps = 50;
encoder_param.audioEncodedCodecType = 0;

TRTCStreamMixingConfig config;
TRTCVideoLayout* video_layout_list = new TRTCVideoLayout[3];

TRTCUser* fixedVideoUser0 = new TRTCUser();
fixedVideoUser0->intRoomId = 1234;
fixedVideoUser0->userId = "mike";
video_layout_list[0].zOrder = 0;
video_layout_list[0].rect.left = 0;
video_layout_list[0].rect.top = 0;
video_layout_list[0].rect.right = 720;
video_layout_list[0].rect.bottom = 1280;
video_layout_list[0].fixedVideoStreamType =
      TRTCVideoStreamType::TRTCVideoStreamTypeSub;
video_layout_list[0].fixedVideoUser = fixedVideoUser0;

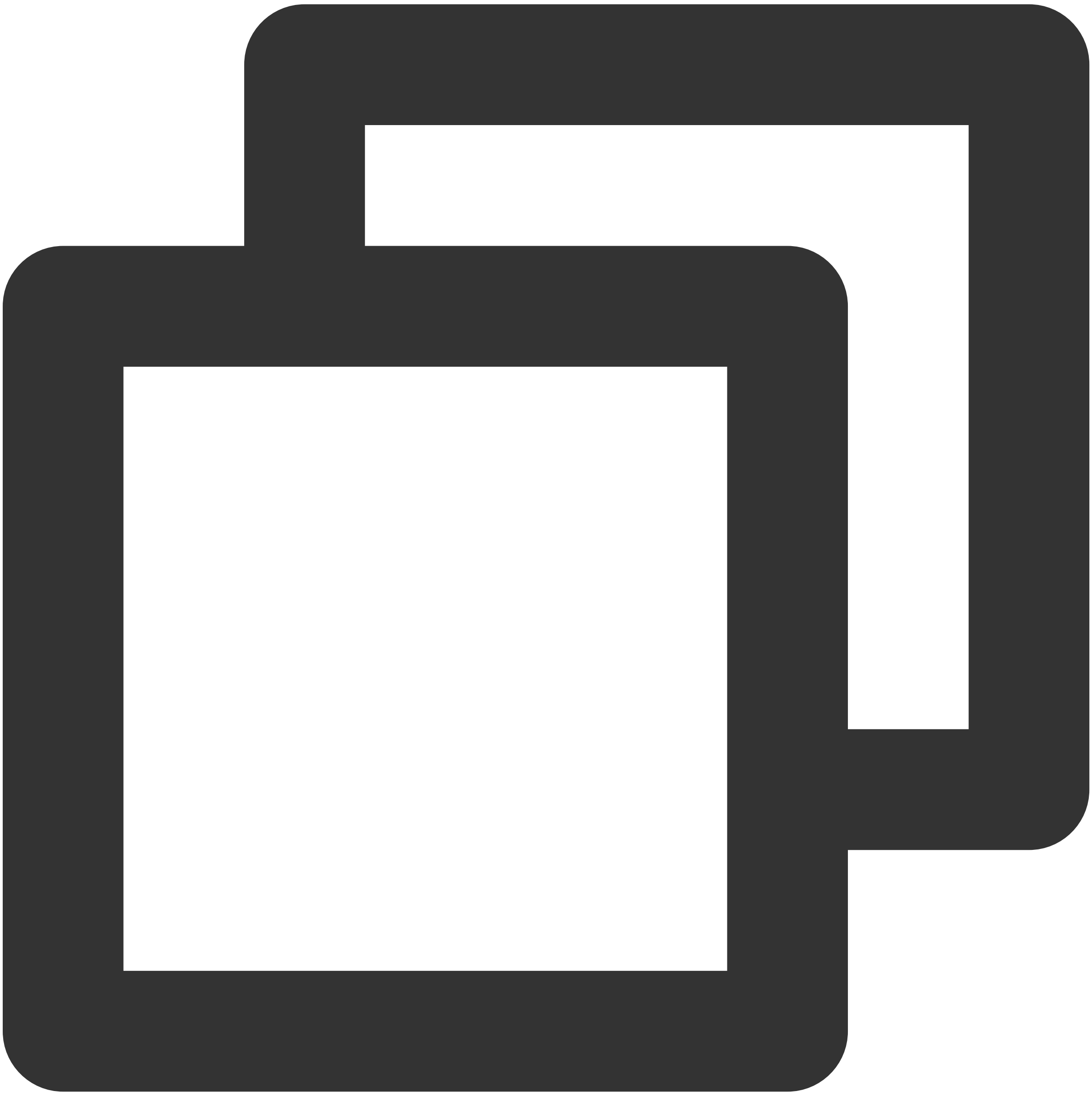
TRTCUser* fixedVideoUser1 = new TRTCUser();
fixedVideoUser1->intRoomId = 1234;
fixedVideoUser1->userId = "mike";
video_layout_list[1].zOrder = 0;
video_layout_list[1].rect.left = 1300;
video_layout_list[1].rect.top = 0;
video_layout_list[1].rect.right = 300;
video_layout_list[1].rect.bottom = 200;
video_layout_list[1].fixedVideoStreamType =
      TRTCVideoStreamType::TRTCVideoStreamTypeBig;
video_layout_list[1].fixedVideoUser = fixedVideoUser1;

video_layout_list[2].zOrder = 0;
video_layout_list[2].rect.left = 1300;
video_layout_list[2].rect.top = 220;
video_layout_list[2].rect.right = 300;
video_layout_list[2].rect.bottom = 200;
video_layout_list[2].fixedVideoStreamType =
      TRTCVideoStreamType::TRTCVideoStreamTypeSub;
video_layout_list[2].fixedVideoUser = nullptr;

config.videoLayoutList = video_layout_list;
```

```
config.videoLayoutListSize = 3;
config.audioMixUserList = nullptr;

trtc->startPublishMediaStream(&target, &encoder_param, &config);
delete fixedVideoUser0;
delete fixedVideoUser1;
delete[] video_layout_list;
```



```
TRTCPublishTarget target = TRTCPublishTarget();
target.mode = TRTCPublishMode.TRTCPublishMixStreamToCdn;
```

```
TRTCPublishCdnUrl cdnUrlEntity = new TRTCPublishCdnUrl();
cdnUrlEntity.rtmpUrl = "rtmp://tencent/live/bestnews";
cdnUrlEntity.isInternalLine = true;
target.cdnUrlList.add(cdnUrlEntity);

TRTCStreamMixingConfig config = TRTCStreamMixingConfig();
TRTCUser selfUser = TRTCUser();
selfUser.userId = localUserId;
selfUser.intRoomId = localRoomId;

TRTCVideoLayout selfVideoLayout = TRTCVideoLayout();
selfVideoLayout.fixedVideoStreamType = TRTCVideoStreamType.TRTCVideoStreamTypeBig;
selfVideoLayout.rect = Rect(originX: 0, originY: 0, sizeWidth: 1080, sizeHeight: 19
selfVideoLayout.zOrder = 0;
selfVideoLayout.fixedVideoUser = selfUser;
selfVideoLayout.fillMode = TRTCVideoFillMode.TRTCVideoFillMode_Fit;
config.videoLayoutList.add(selfVideoLayout);TRTCUser remoteUser = TRTCUser();

remoteUser.userId = remoteUserId;
remoteUser.intRoomId = remoteRoomId;
TRTCVideoLayout remoteVideoLayout = TRTCVideoLayout();
remoteVideoLayout.fixedVideoStreamType = TRTCVideoStreamType.TRTCVideoStreamTypeBig
remoteVideoLayout.rect = Rect(originX: 100, originY: 50, sizeWidth: 216, sizeHeight
remoteVideoLayout.zOrder = 1;
remoteVideoLayout.fixedVideoUser = remoteUser;
remoteVideoLayout.fillMode = TRTCVideoFillMode.TRTCVideoFillMode_Fit;
config.videoLayoutList.add(remoteVideoLayout);

TRTCStreamEncoderParam param = TRTCStreamEncoderParam();
param.videoEncodedWidth = 1080;
param.videoEncodedHeight = 1920;
param.videoEncodedKbps = 5000;
param.videoEncodedFPS = 30;
param.videoEncodedGOP = 3;
param.audioEncodedSampleRate = 48000;
param.audioEncodedChannelNum = 2;
param.audioEncodedKbps = 128;
param.audioEncodedCodecType = 2;

trtcCloud.startPublishMediaStream(target: target, config: config, params: param);
```

FAQs

1. Can I listen for the status of CDN streams? What should I do if an error occurs?
You can listen for the `onCdnStreamStateChanged` callback event to get the latest status of a relay to CDN task.
2. How do I switch from publishing a single stream to publishing mixed streams? Do I need to stop publishing first and create a new relay task?
To switch from publishing a single stream to publishing mixed streams, just call `updatePublishMediaStream` , passing in the `taskid` of the current task. Note that, in order to ensure the stability of the publishing process, you cannot switch from publishing a single stream to mixing and publishing only audios or only videos. By default, both audio and video data are published when you publish a single stream. If you switch to publishing mixed streams, you must also publish both audios and videos.
3. How can I mix only videos (without audio)?
Do not set the audio parameters in `TRTCStreamEncodeParam` and leave `audioMixUserList` of `TRTCStreamMixingConfig` empty.
4. Can I add watermarks to mixed streams?
Yes, you can use `watermarkList` of `TRTCStreamMixingConfig` to set watermarks.
5. In online learning scenarios, can I mix the screen shared by the teacher?
Yes, you can. We recommend you publish the screen as the substream and mix the teacher's camera video and screen. When specifying the stream mixing parameters, set `fixedVideoStreamType` of `TRTCVideoLayout` to `TRTCVideoStreamTypeSub` .
6. When a preset layout is used, how are audios mixed?
When a preset layout is used, TRTC will mix the audios of up to 16 users in the room.

Billing

Cost calculation

Relay to CDN fees are charged based on the peak bandwidth used each month. If you mix streams, the MCU cluster will decode and re-encode the streams in the cloud, so an additional transcoding fee will be charged. Transcoding fees vary with the resolutions of the streams transcoded and the transcoding duration. For details, see Billing of MixTranscoding and Relay to CDN.

Cost control

If you use client-side APIs, stream mixing stops at the backend when either of the following conditions is met.
The anchor who called `startPublishMediaStream` to mix streams left the room.
The anchor called `stopPublishMediaStream` to stop mixing the streams.
In all other cases, TRTC will continue to mix streams in the cloud. Therefore, to reduce costs, when you no longer want to mix streams, please stop it using either of the above methods.

Enabling Advanced Permission Control

Last updated：2024-08-09 22:25:01

Overview

You may consider **enabling Advanced Permission Control** if you want to allow only specific users to enter a room or use their mics, but are worried that giving permissions on the client side makes the service vulnerable to attacks and cracking.

You do not need to enable advanced permission control in the following scenarios:

Scenario 1: You want an audience as large as possible and do not want to control access to rooms.

Scenario 2: Preventing client-side attacks is not your priority at the moment.

We recommend that you enable advanced permission control for enhanced security in the following scenarios:

Scenario 1: Your video or audio calls have high security requirements.

Scenario 2: You want to implement different access controls for different rooms.

Scenario 3: You want to control the use of mics by audience.

Supported Platforms

iOS	Android	macOS	Windows	Electron	Web	Flutter
✓	✓	✓	✓	✓	✓	✓

Understanding Advanced Permission Control

After you enable advanced permission control, TRTC will verify not only `UserSig` (the room entry ticket), but also **PrivateMapKey** (the permission ticket). The latter contains an encrypted `roomid` and permission bit list.

A user providing only `UserSig` but not `PrivateMapKey` will be unable to enter the specified room.

The permission bit list in `PrivateMapKey` uses the eight bits of a byte to represent different permissions for users holding `PrivateMapKey` .

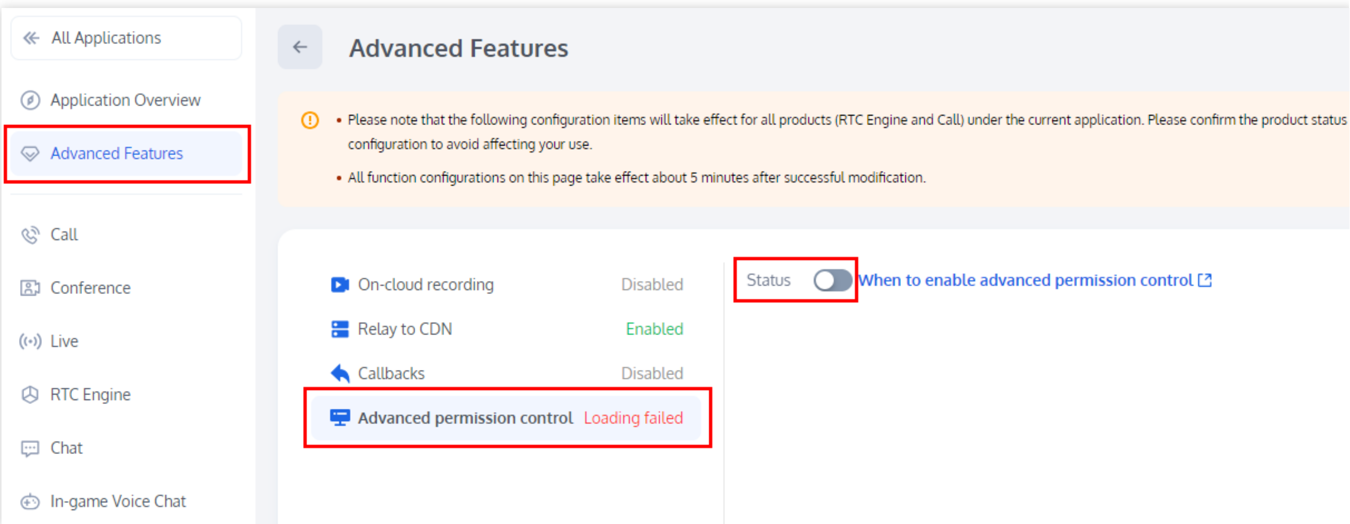
Bit Sequence	Binary	Decimal	Permission
First	0000 0001	1	Room creation
Second	0000 0010	2	Room entry
Third	0000 0100	4	Sending audio

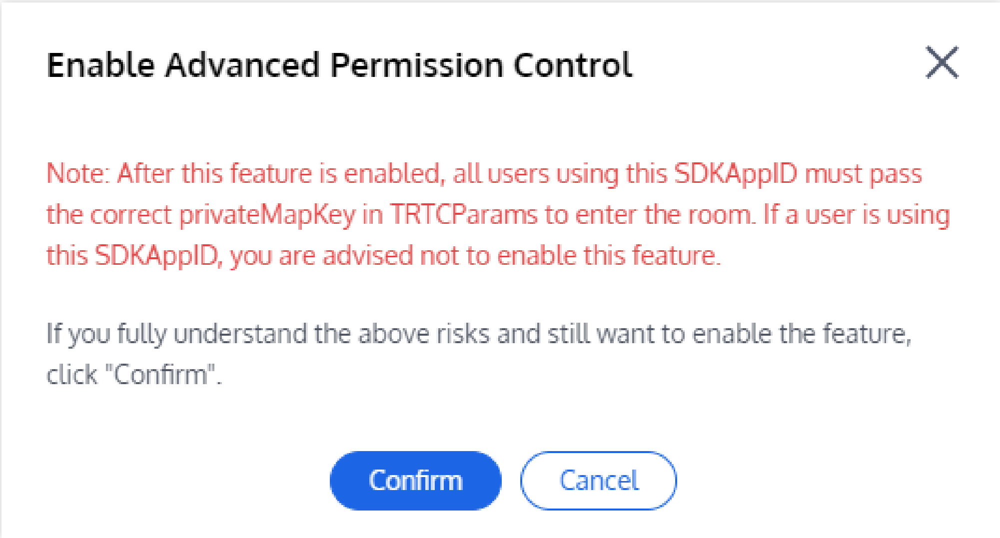
Fourth	0000 1000	8	Receiving audio
Fifth	0001 0000	16	Sending video
Sixth	0010 0000	32	Receiving video
Seventh	0100 0000	64	Sending substream (screen sharing) video
Eighth	1000 0000	128	Receiving substream (screen sharing) video

Enabling Advanced Permission Control

Step 1. Log in to the TRTC console and enable advanced permission control

1. Log to Tencent RTC Console > Applications, click on **Manage** in the row of the target application whose feature configuration needs to be modified, and select **Advanced Features** from the project column on the left.
2. In **Advanced Features**, click the button on the right side of **Enable Advanced Permission Control**, and in the pop-up window, click **Confirm** to complete the activation.





Note：

After you enable advanced permission control for an application (`SDKAppid`), all users using the application must pass `privateMapKey` in `TRTCParams` to enter a room (as described in Step 2 below). Therefore, you are not advised to enable the feature if you have active users using the application.

Step 2. Calculate `PrivateMapKey` on your server

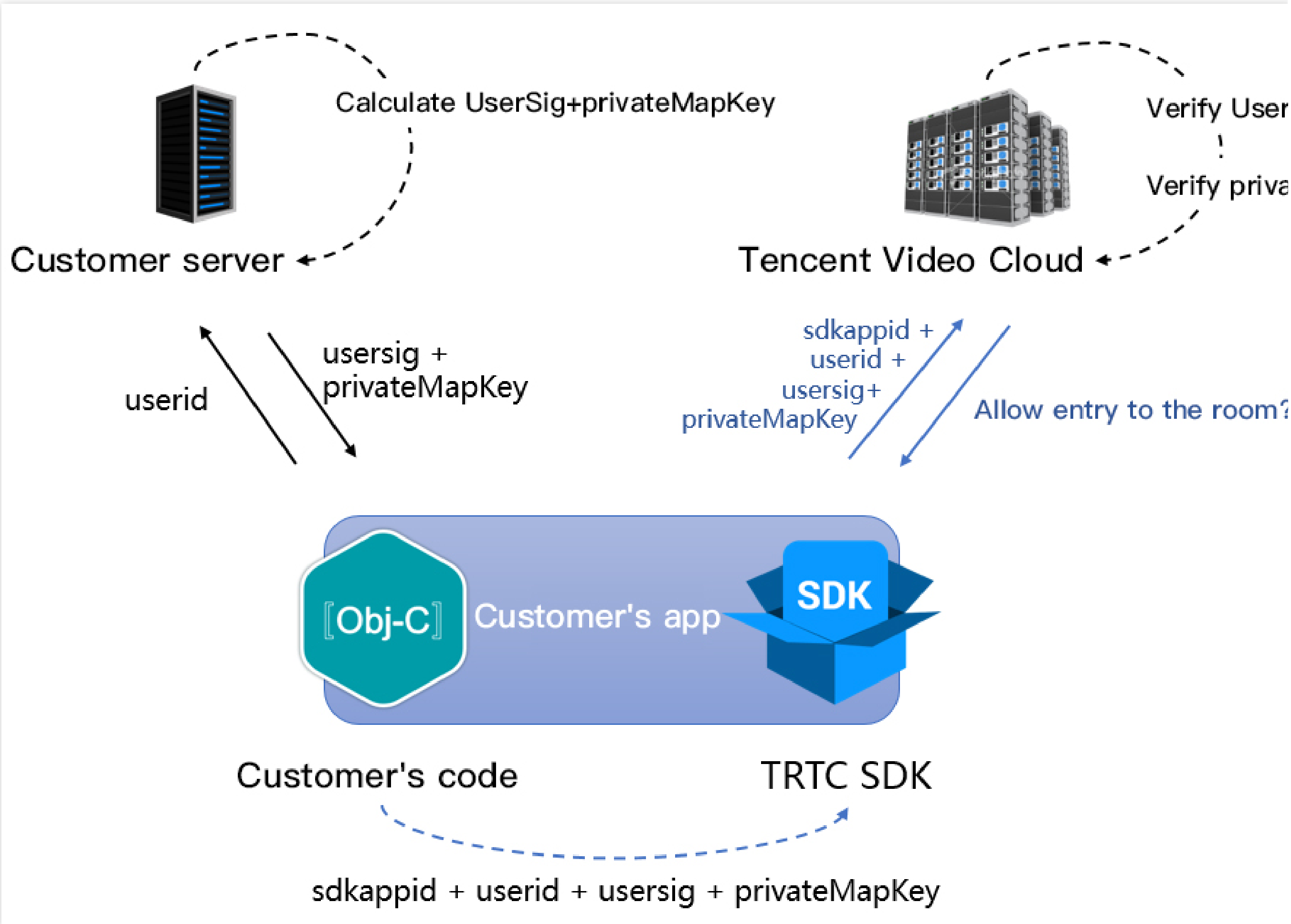
`PrivateMapKey` protects the client from being reverse engineered and cracked and consequently prevents non-members from entering high-level rooms. Therefore, instead of calculating `PrivateMapKey` directly on your application, you should do so on your server and then return the result to your application.

We provide `PrivateMapKey` calculation codes for Java, GO, PHP, Node.js. Python, C#, and C++. You can download and integrate them into your server.

Programming Language	Key Functions	Download Link
Java	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub
GO	`GenPrivateMapKey` and `GenPrivateMapKeyWithStringRoomID`	GitHub
PHP	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub
Node.js	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub
Python	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub

C#	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub
C++	`genPrivateMapKey` and `genPrivateMapKeyWithStringRoomID`	GitHub

Step 3. Distribute `PrivateMapKey` from your server to your application



As shown in the figure above, `PrivateMapKey` is calculated on your server and distributed to your application, which can then pass the `PrivateMapKey` to the SDK via two methods.

Method 1: passing `PrivateMapKey` to the SDK when calling `enterRoom`

You can set **privateMapKey** in TRTCParams when calling the `enterRoom` API of `TRTCCloud`.
This method verifies `PrivateMapKey` when users enter a room. It is simple and is used to assign permissions to users before room entry.

Method 2: updating `PrivateMapKey` to the SDK through an experimental API

During live streaming, when audience turn their mics on to co-anchor, TRTC will re-verify the `PrivateMapKey` carried in `TRTCParams` at the time of room entry. That means if you set a short validity period for `PrivateMapKey` , such as 5 minutes, the re-verification may fail and cause the audience to be removed from the room when they switch to the role of “anchor”.
To solve this issue, you can extend the validity period, for example, from 5 minutes to 6 hours or, before the audience call `switchRole` to switch to the role of “anchor”, apply for a new `PrivateMapKey` from your server and update it to the SDK by calling the experimental API `updatePrivateMapKey` . Below is the sample code:

Android
iOS
C++
C#


```
JSONObject jsonObject = new JSONObject();
try {
    jsonObject.put("api", "updatePrivateMapKey");
    JSONObject params = new JSONObject();
    params.put("privateMapKey", "xxxxx"); // Enter the new `privateMapKey`.
    jsonObject.put("params", params);
    mTRTCCloud.callExperimentalAPI(jsonObject.toString());
} catch (JSONException e) {
    e.printStackTrace();
}
```

```
NSMutableDictionary *params = [[NSMutableDictionary alloc] init];
[params setObject:@"xxxxx" forKey:@"privateMapKey"]; // Enter the new `privateMapKe
NSDictionary *dic = @{@"api": @"updatePrivateMapKey", @"params": params};
NSData *jsonData = [NSJSONSerialization dataWithJSONObject:dic options:0 error:NULL
NSString *jsonStr = [[NSString alloc] initWithData:jsonData encoding:NSUTF8StringEn
[WXTRTCCloud sharedInstance] callExperimentalAPI:jsonStr];
```

```
std::string api = "{\\"api\\":\\"updatePrivateMapKey\\",\\"params\\":{\\"privateMap
TRTCCloudCore::GetInstance()->getTRTCCloud()->callExperimentalAPI(api.c_str());
```

```
std::string api = "{\\"api\\":\\"updatePrivateMapKey\\",\\"params\\":{\\"privateMap
mTRTCCloud.callExperimentalAPI(api);
```

FAQs

1. Why can't I enter any online room?

After you enable room permission control for an application (`SDKAppid`), users must pass `PrivateMapKey` in `TRTCParams` to enter any room under the application. Therefore, if your online business is running, and you haven't integrated into it the `privateMapKey` logic, please do not enable room permission control.

2. What is the difference between `PrivateMapKey` and `UserSig` ?

`UserSig` is a required parameter of `TRTCParams` , which is used to check whether the current user is authorized to use TRTC services and prevent attackers from stealing the traffic in your application (`SDKAppid`). `PrivateMapKey` is an optional parameter of `TRTCParams` , which is used to check whether the current user is authorized to enter the specified room (`roomid`) and confirm the user's permissions in the room. Use `PrivateMapKey` only if you need to distinguish users from one another.

Push Media Stream into TRTC

Last updated：2024-08-12 17:57:06

Overview

Watch together, listen together, play together, learn together... Various experiences that once required face-to-face interaction are now moving online. Even if separated by thousands of miles, friends can still watch movies, listen to music and chat together. This amazing real-time interactive experience is becoming popular among young people and is now a major feature and mainstream direction of audio and video products.

TRTC offers two streaming solutions: push online media stream and RTMP streaming with TRTC, each with their own application scenario, as detailed below:

Push online media stream is used to **pull cloud-based online media streams (online streaming or cloud-based on-demand files)** and push them into a TRTC room.

RTMP streaming with TRTC is used to **stream local media files and audio and video from capture devices** into a TRTC room via the RTMP standard protocol.

Note:

Relevant fees are as follows:

Feature unlocking: **Push online media stream** and **RTMP streaming with TRTC** features need to be unlocked by a subscription to the **basic** or **professional version** of RTC-Engine monthly package.

Usage fees:

Using the streaming feature involves transcoding operations, incurring transcoding costs. For more details, see the Description of Billing of MixTranscoding and Bypass Relay.

The costs of audio duration incurred by the streaming robot in the room are charged (Note: The costs incurred by the robot in the room for push online media stream features will not be charged by August 15, 2024, and they will begin to be charged on August 16, 2024).

The audience in the room subscribing to audio and video streams will incur audio and video call costs. For details, see the Description of Billing of Audio and Video Duration.

Push Online Media Stream

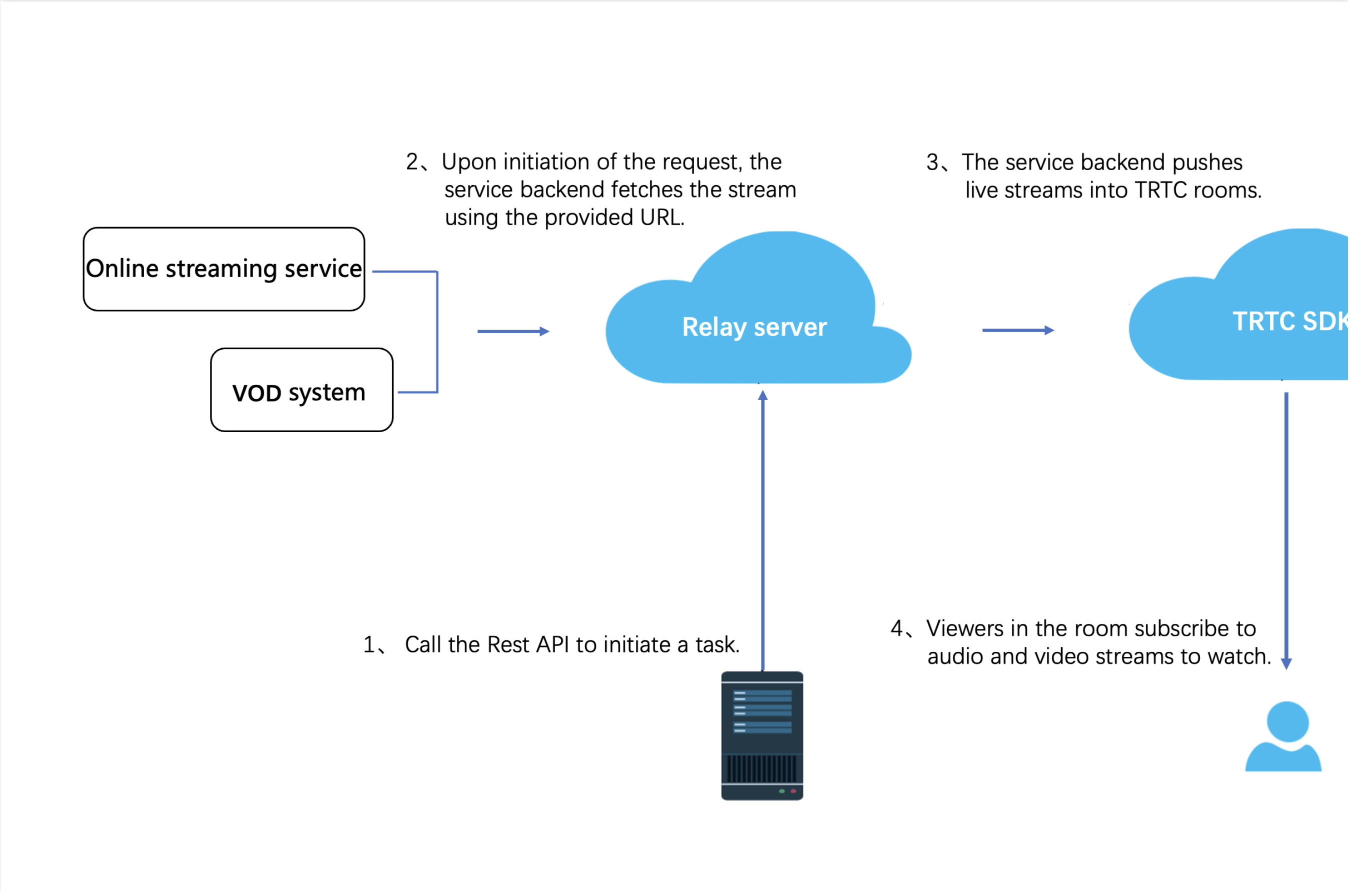
Application Scenario

Scenario type	Description
AI interactive classroom	Relying on TRTC's ability to push online media stream, the platform enables online live interactive teaching by combining recorded real-life teaching videos with AI technology. This significantly reduces operational costs while ensuring teaching effectiveness. Before class,

	the platform records video segments for explanation of knowledge points, interactive questioning, feedback on questions, and answers based on the teacher's course setup, and uploads them to the video library. During the class, these videos are streamed to the TRTC room for live broadcasting through the TRTC push online media capability. Students can engage in interactive learning through voice and touchscreen. The server uses AI technology to evaluate real-time voice and responses of students, seamlessly switches teaching segments, and provides different real-time feedback, thus offering a personalized teaching experience.
"Watch together" room service	Live streaming contents, such as game live streaming, fashion shows, and sports events, can be pushed to a TRTC room through the TRTC's push online media stream capability for ultra-low latency synchronized viewing within the room. With TRTC's real-time interaction capability, the audience can communicate in real-time, cheer together, and enjoy an immersive viewing experience. On-demand programs like movies and music can also be inputted into the TRTC room through the capability, allowing users to share in real-time and chat with friends while watching.

Feature Architecture

1. Users create push online media stream tasks using the REST API. These tasks are executed by the relay server.
2. The relay server pulls online streams or on-demand files.
3. The relay server pushes the fetched audio and video to the TRTC room and automatically generates a virtual anchor user. The username and room number of this user are specified when the task is created.
4. Other TRTC clients can watch these streams and utilize TRTC capabilities such as recording and relay.



Feature Description

The feature description of push online media stream is as follows:

Type	Description
Task initiation method	Users can initiate push online media stream tasks via the REST API. The audience can watch these streams, and features such as recording and relay are supported.
Multiple source protocols and formats	Protocols: HTTP, HTTPS, RTMP, HLS Formats: FLV, MP3, MP4, MPEG-TS, MOV, MKV, M4A Video encoding: H.264, VP8 Audio encoding: AAC, OPUS
Server-side callback	When a push online media stream task is created and completed, it can be called backed to the server on the service side for service logic purposes. For detailed push online media stream events, go to view.

Related Rest API

Start push online media stream: StartStreamIngest

Stop push online media stream: StopStreamIngest

Query push online media stream: DescribeStreamIngest

RTMP Streaming With TRTC

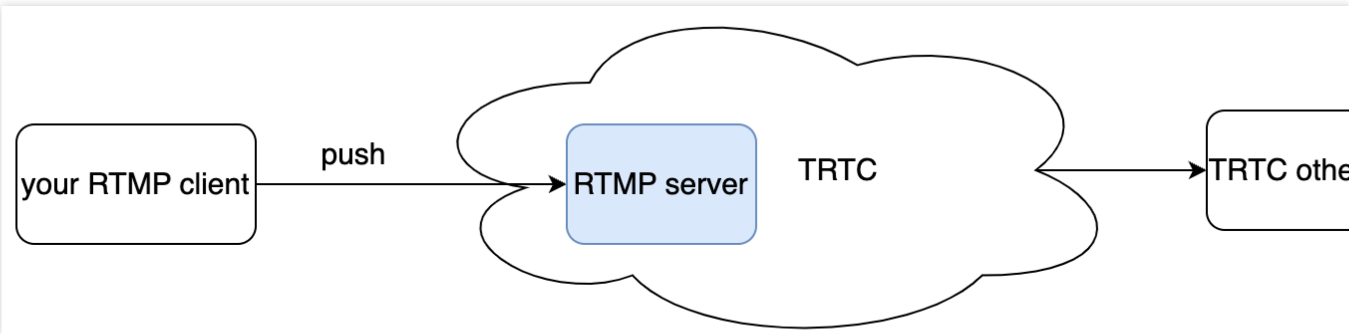
TRTC supports the streaming of **local media files** and **audio and video from capture devices** into a TRTC room via the RTMP standard protocol. To facilitate your integration of TRTC, you can install OBS, FFmpeg, or other RTMP libraries to realize streaming with TRTC. OBS is a third-party open-source tool for live streaming. It is easy to use and free of charge, and it supports OS X, Windows, and Linux. OBS can be used in a wide range of scenarios and is capable of meeting most live streaming needs without requiring additional plugins. You can download its latest version from the OBS website.

Use Cases

Scenario	Description
Online education	Use the desktop edition of OBS or FFmpeg to publish learning materials (most media formats are supported) over RTMP to a TRTC room. Students in the room can play the stream via the TRTC SDK and see the same learning materials as the teacher controls the playback progress/speed or switches between chapters. Excellent synchronization across multiple devices ensures better teaching quality.
Sports watching	Sports event organizers provide content in the form of RTMP streams. You can publish the streams to TRTC rooms so that users in the rooms can watch the event with ultra-low latency. With TRTC's interaction capability, users can also audio/video chat with each other throughout an event.
Others	You can also use the RTMP publishing feature to implement other real-time interactive applications based on streaming.

Architecture

An RTMP client is a module of TRTC and can communicate with other TRTC clients. The interconnection delay is less than 600ms under normal circumstances. It can also use TRTC capabilities such as recording and relaying. The network architecture is shown in the figure below.



Publishing and Playback URLs

Publishing URLs

```
rtmp://intl-rtmp.rtc.qq.com/push/Room ID?sdkappid=Application ID&userid=User ID&use
```

For RTMP publishing, `appName` is `push` .

Replace "Room ID", "Application ID", "User ID", and "Signature" with their actual values.

For the sake of simplicity, we support only string-type room IDs. A room ID can contain numbers, letters, and underscores and cannot exceed 64 characters.

Warning:

To play an RTMP stream on other TRTC clients, when entering the room, make sure you use a string-type room ID.

For how to generate `UserSig` , see UserSig. **Make sure your signature is within the validity period**.

Example:

```
rtmp://intl-rtmp.rtc.qq.com/push/hello-string-room?sdkappid=140*****66&userid=*****
```

Usage Example

You can use software or a programming library that supports RTMP to publish RTMP streams. The section below shows you how to do this.

Using OBS to publish streams

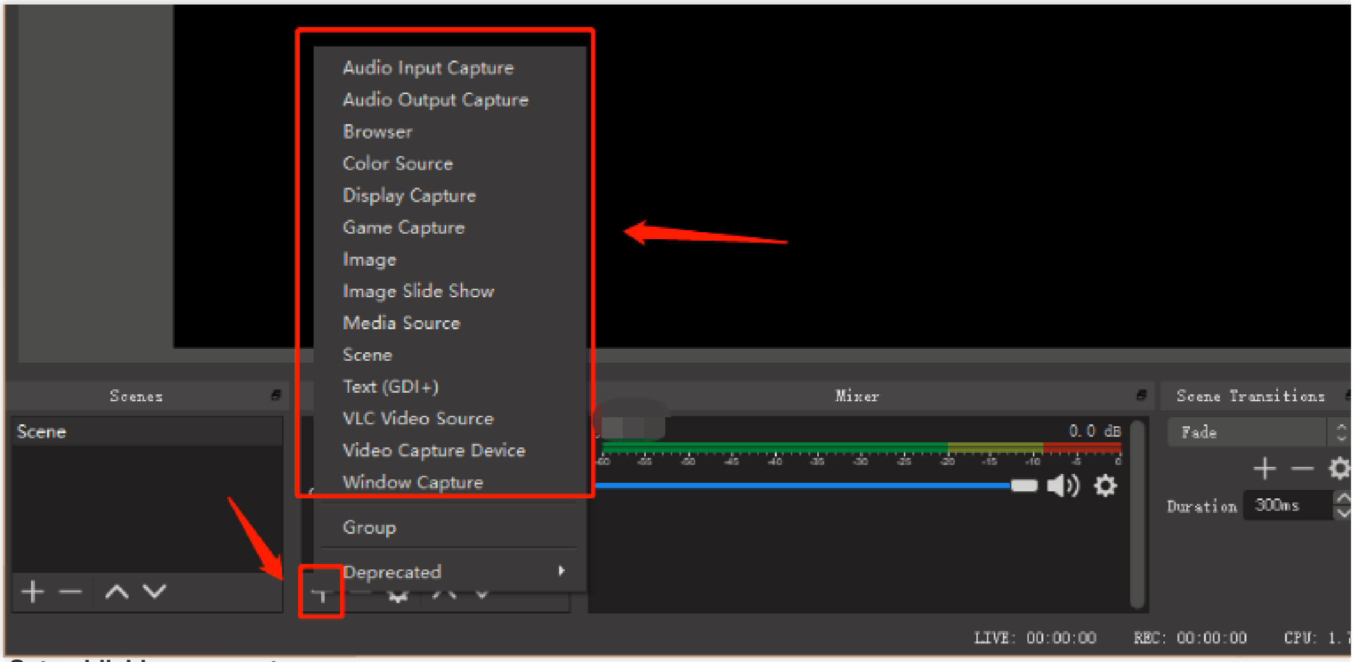
Prerequisites

You have installed OBS.

Step 1. Select a source

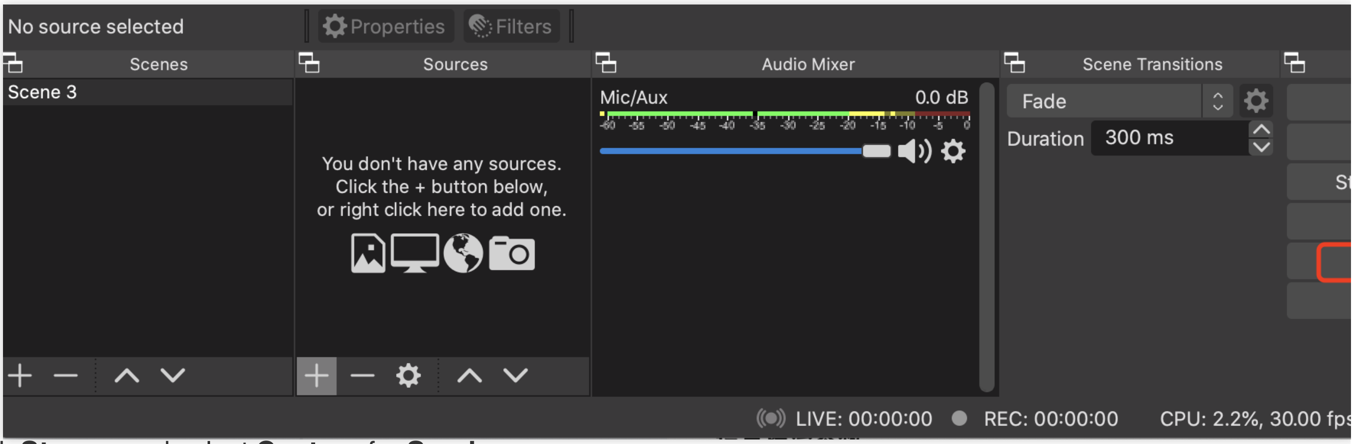
In the **Sources** panel at the bottom, click **+**, and select a source based on your needs. Common sources include the following:

Source	Note
Image	Publishes a single image
Image Slide Show	Publishes multiple images (you can determine the order of playback and whether to loop the playback)
Scene	Inserts an entire scene to enable various streaming effects
Media Source	Uploads a local file and publishes it as a live stream
Text	Adds real-time text to your stream
Window Capture	Captures and publishes the window you select in real time
Video Capture Device	Captures and publishes the images captured by a camera in real time
Audio Input Capture	Audio live streaming (audio input device)
Audio Output Capture	Audio live streaming (audio output device)



Step 2. Set publishing parameters

1. In the **Controls** panel at the bottom, click **Settings**.



2. Click **Stream** and select **Custom** for **Service**.

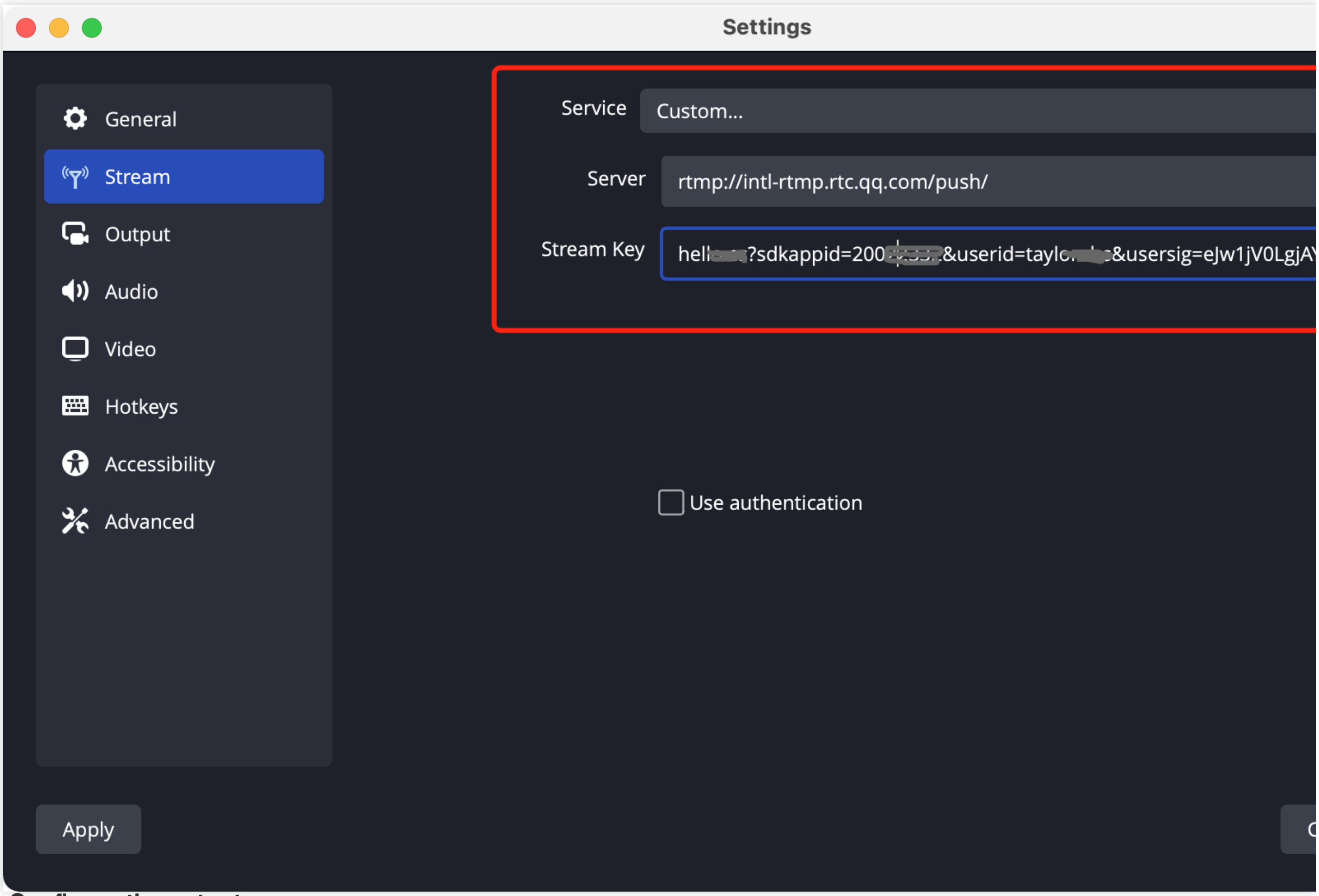
3. Enter `rtmp://intl-rtmp.rtc.qq.com/push/` for **Server**.

4. Enter a stream key in the following format:


```
Room ID?sdkappid=Application&userid=User ID&usersig=Signature
```

Replace "Room ID", "Application ID", "User ID", and "Signature" with the actual values, for example:


```
hello-string-room?sdkappid=140*****66&userid=******rtmp2&usersig=eJw1jdE**********
```



Step 3. Configure the output

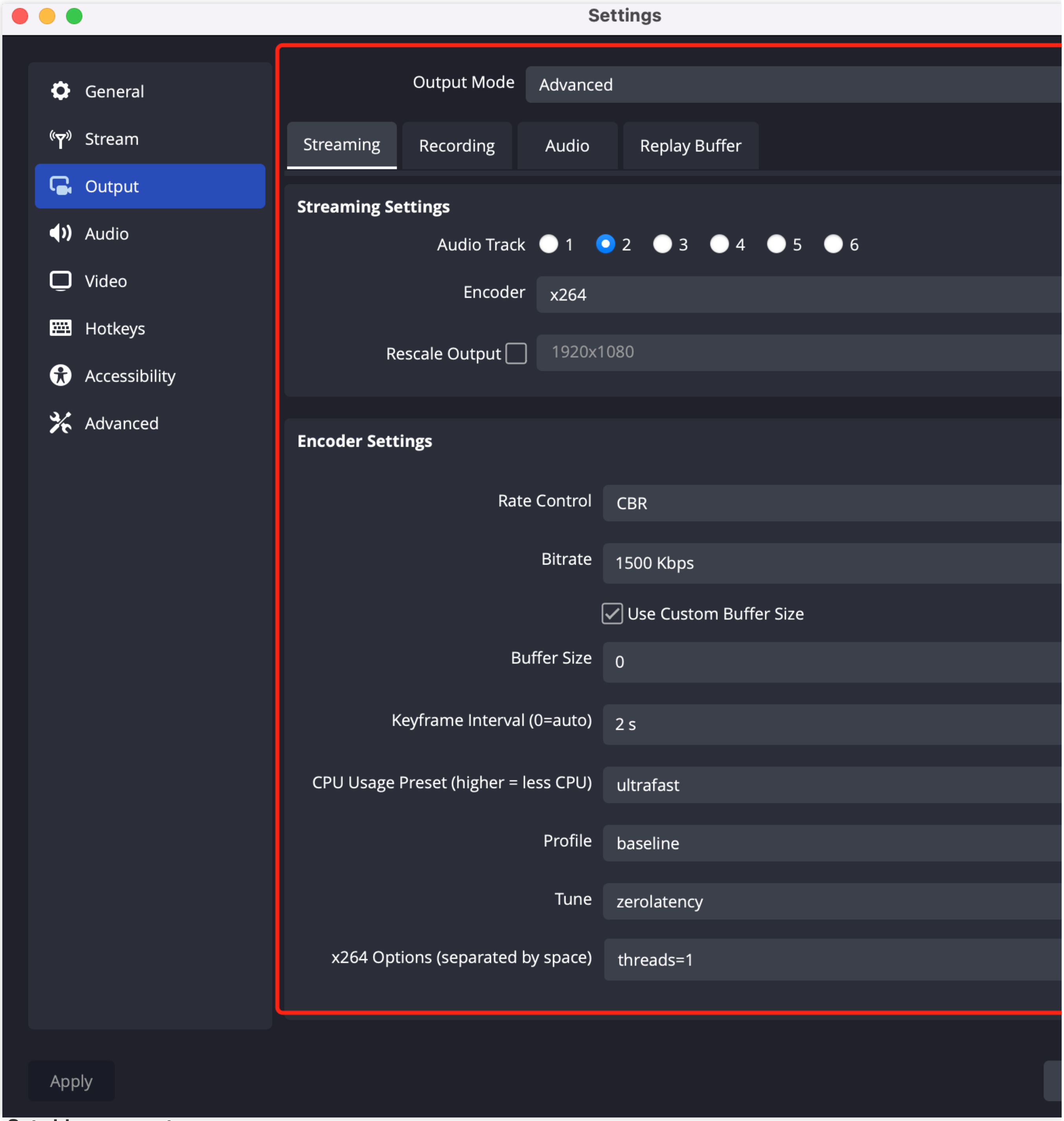
Because RTMP does not support B-frames, set the video encoding parameters as follows to remove B-frames.

1. Go to **Controls** > **Settings** > **Output**.

2. Select **Advanced** for **Output Mode**. The recommended **Keyframe Interval** is **1** or **2**. For **CPU Usage Preset**, select **ultrafast**. For **Profile**, select **baseline**. For **Tune**, select **zerolatency**. And for **x264 Options**, enter `threads=1` , and then click **OK**.

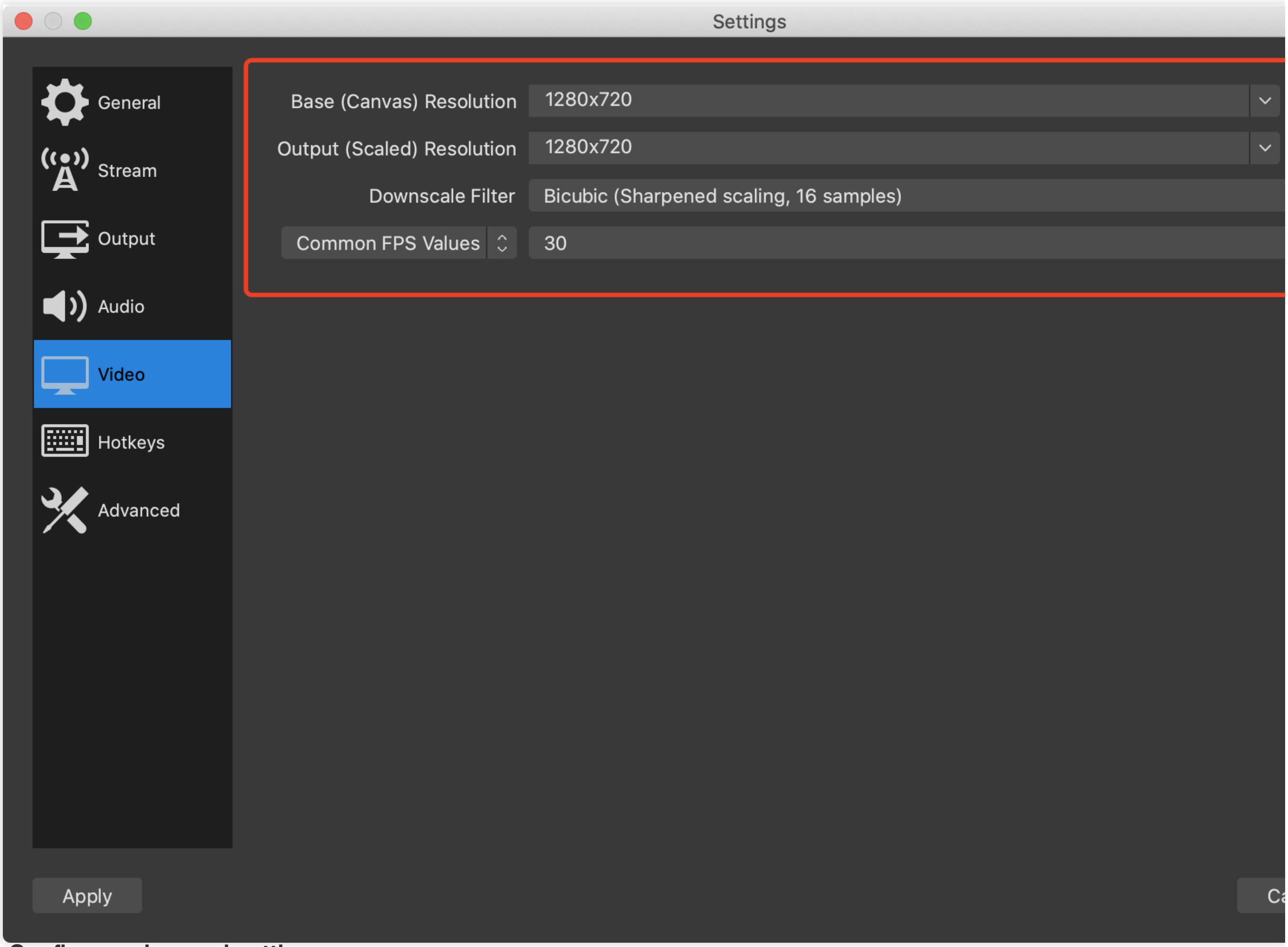
Warning:

You need to remove the B-frames in RTMP streams, otherwise the connection will be disconnected after pushing. To achieve this, select baseline for Profile.



Step 4. Set video parameters

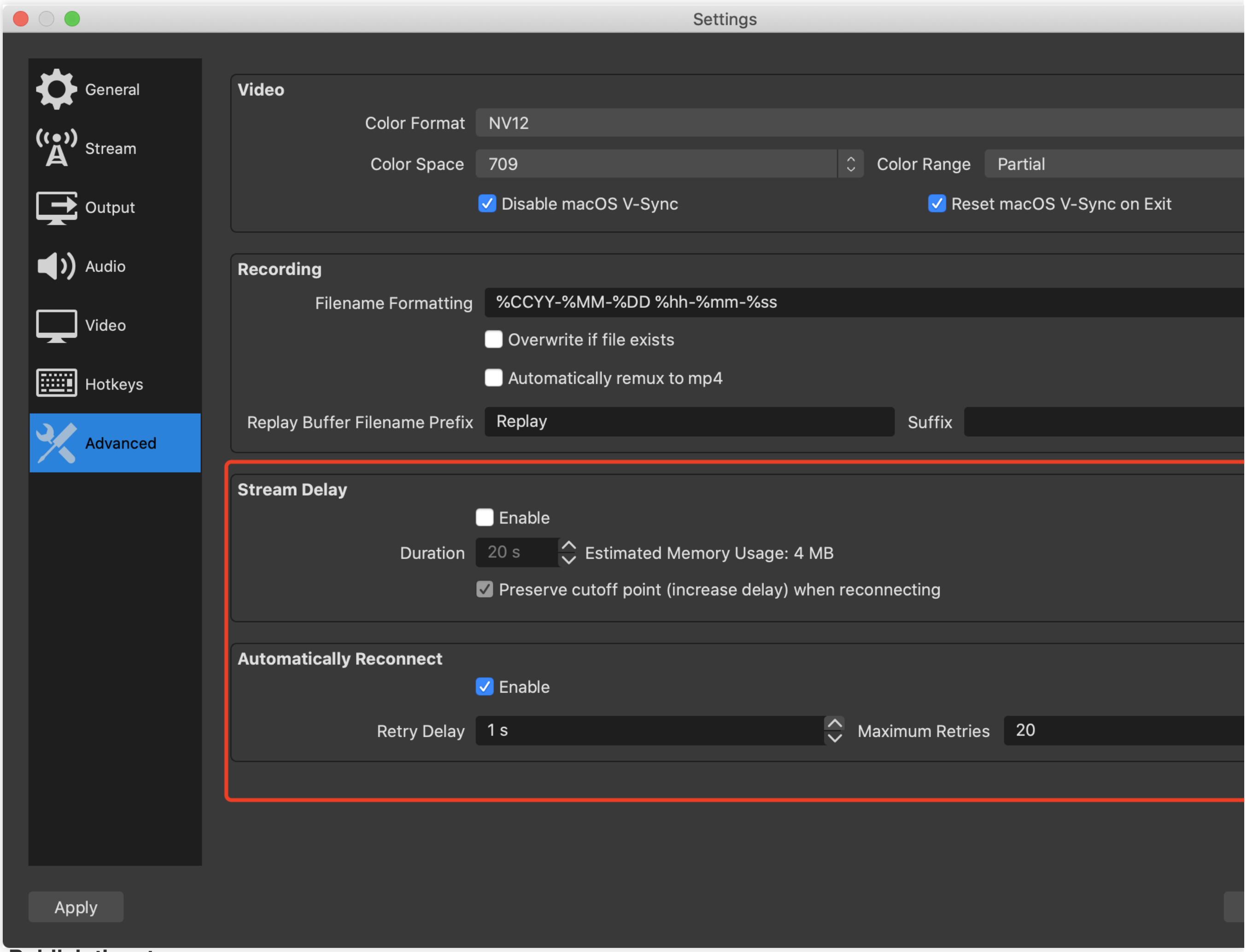
You can set video resolution and frame rate under the **Video** section of **Settings**. Resolution determines the clarity of video shown to audience members. The higher the resolution, the clearer the video. Frame rate (frames per second) determines playback smoothness. Typical frame rate falls in the range of 24 fps to 30 fps. Playback may stutter if frame rate is lower than 16 fps. Video games require higher frame rate and tend to stutter at a frame rate lower than 30 fps.



Step 5. Configure advanced settings

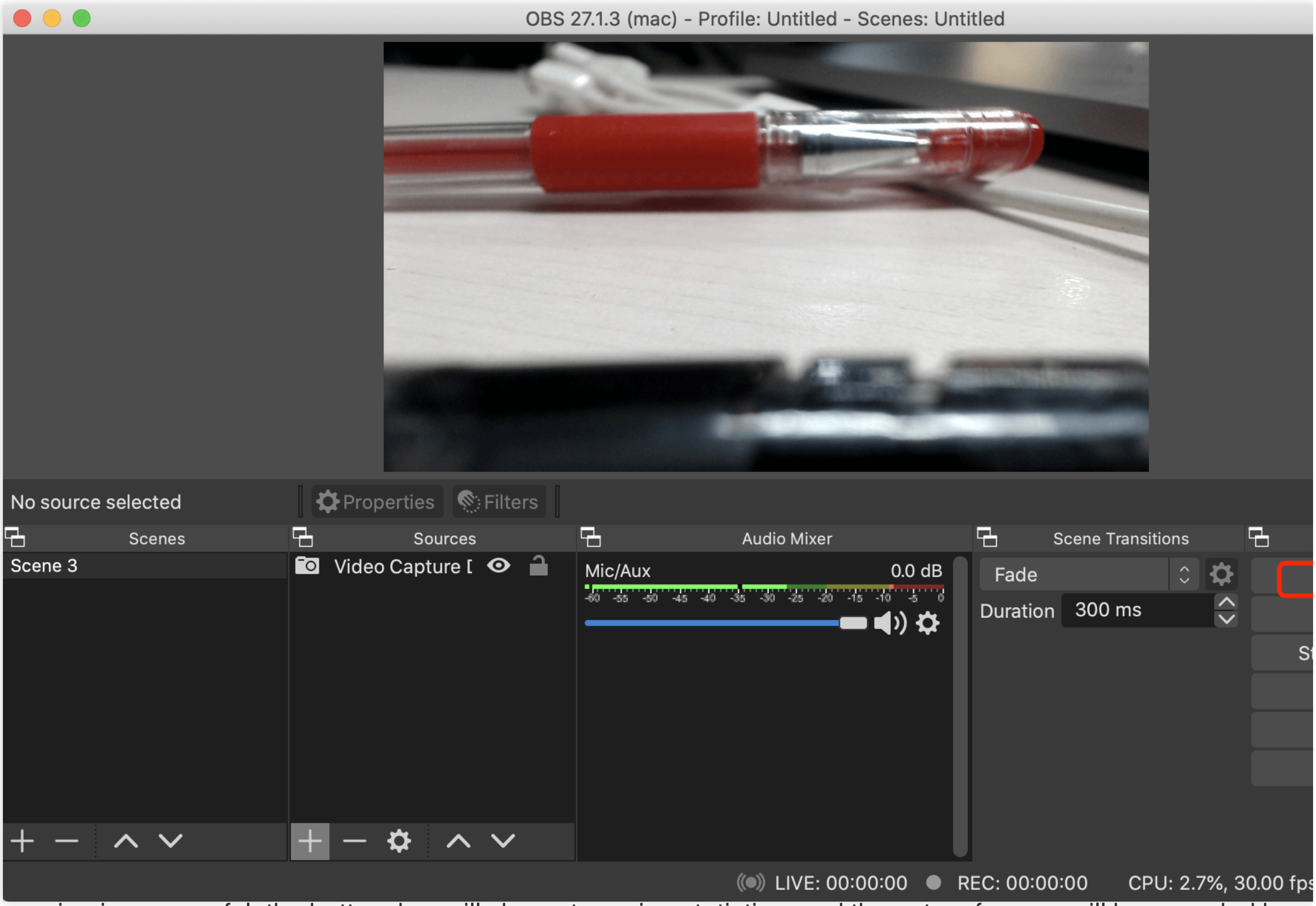
To reduce end-to-end delay, we recommend you do not enable **Stream Delay**.

Keep **Automatically Reconnect** enabled and make **Retry Delay** as short as possible so that the publisher can be reconnected quickly after a disconnection occurs due to network jitter.

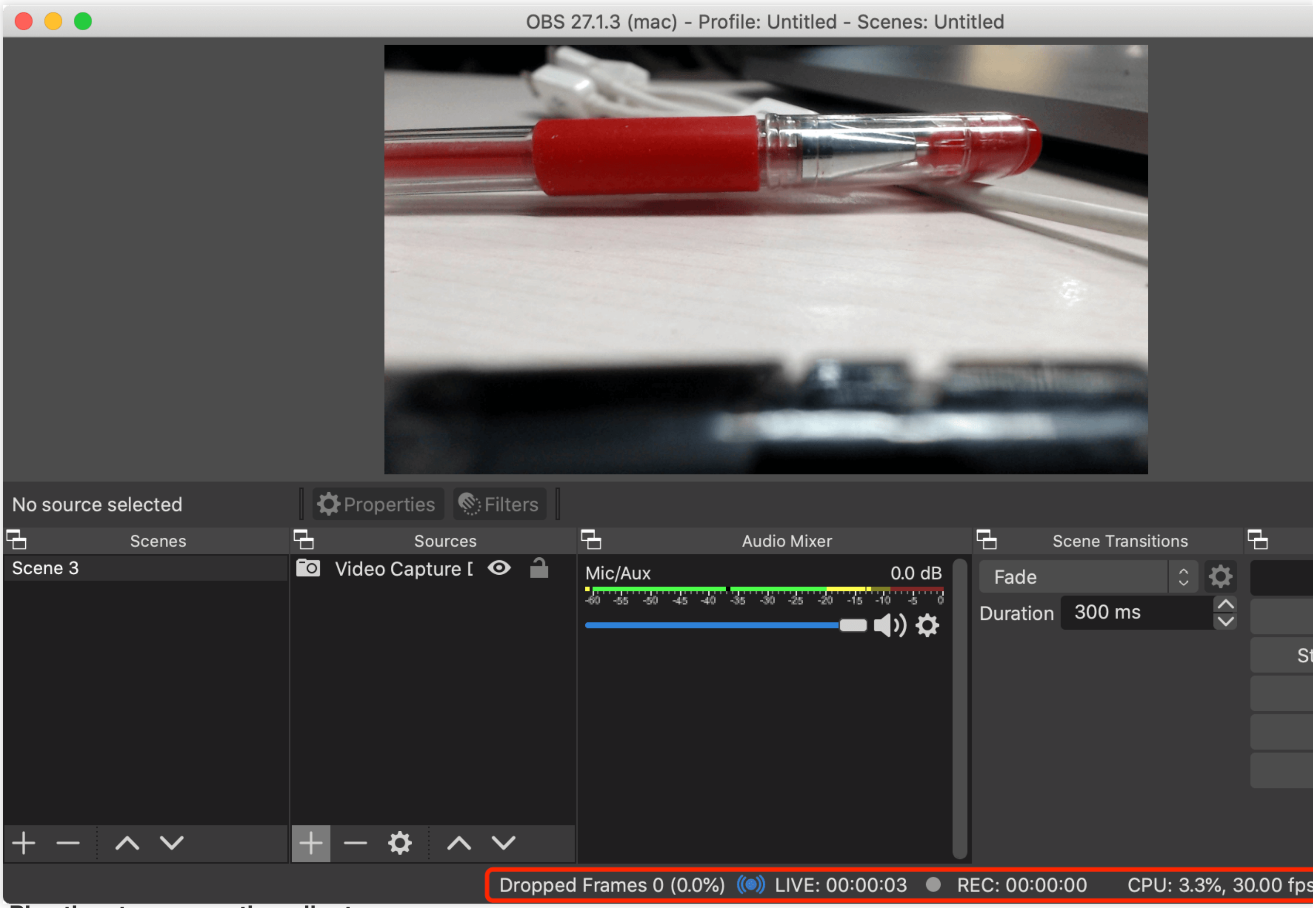


Step 6. Publish the stream

1. In the **Controls** panel at the bottom, click **Start Streaming**.

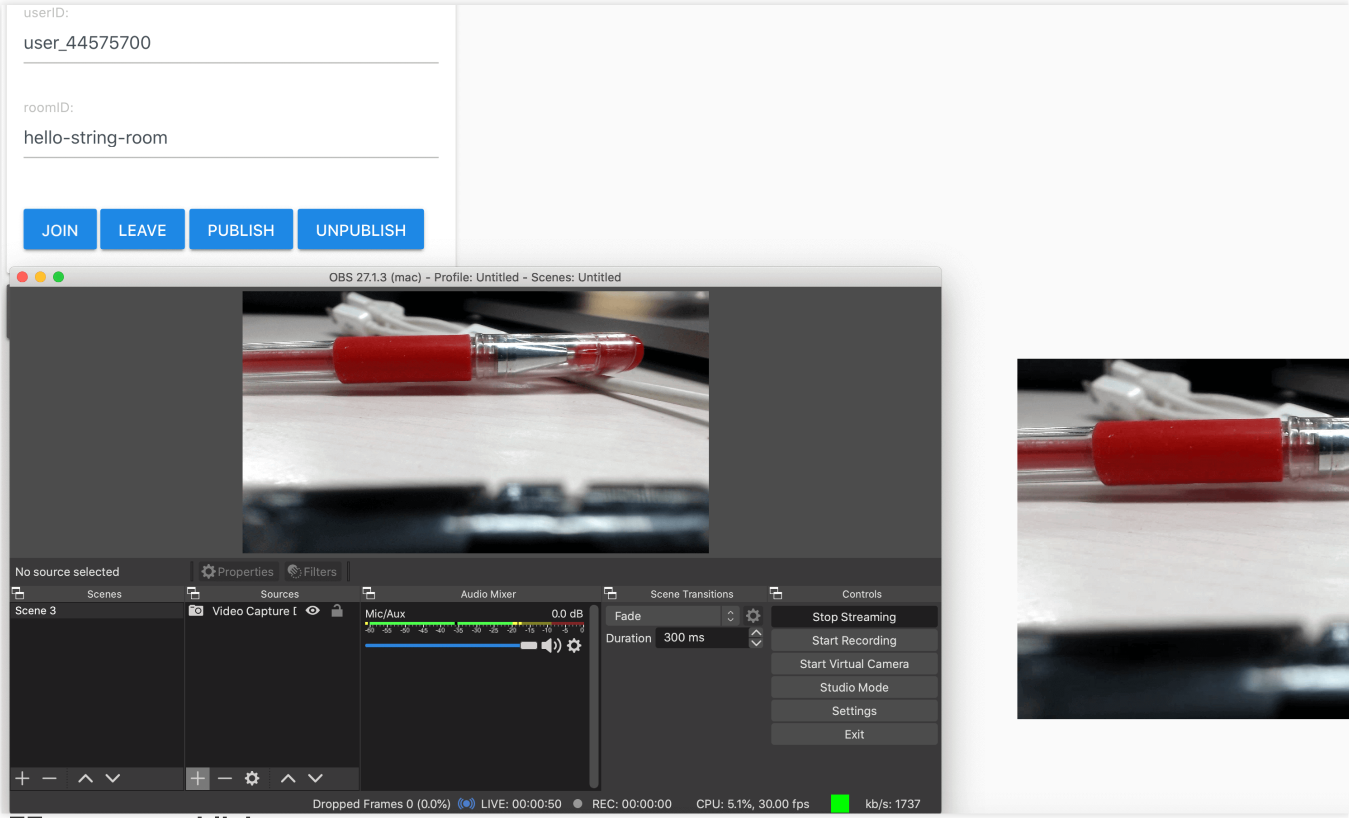


2. If streaming is successful, the bottom bar will show streaming statistics, and the entry of a user will be recorded by the TRTC monitoring dashboard.



Step 7. Play the stream on other clients

As mentioned above in Set publishing parameters, to play the RTMP stream on other TRTC clients, you need to use a string-type room ID when entering the room. The screenshot below is an example of playing the RTMP stream on Web. （ps : You can go to the Demo pageand enter the room on any client-side to view the stream .）



Using FFmpeg to publish streams

To publish RMTP streams using FFmpeg commands or other RTMP libraries, you need to use the full URL, the H.264 video codec, and the AAC audio codec. For the container format, FLV is recommended. For GOP, one or two seconds is recommended.

The configuration of FFmpeg parameters varies with different scenarios, so you need to have some knowledge of FFmpeg in order to use it to publish streams. The table below lists some common FFmpeg commands. For more options, see FFmpeg documentation.

FFmpeg commands

```
ffmpeg [global_options] {[input_file_options] -i input_url} ... {[output_file_optio
```

Common FFmpeg options

Option	Note
-re	Reads input at the native frame rate. This is usually used to read local files.

Options for **output_file_options** include:

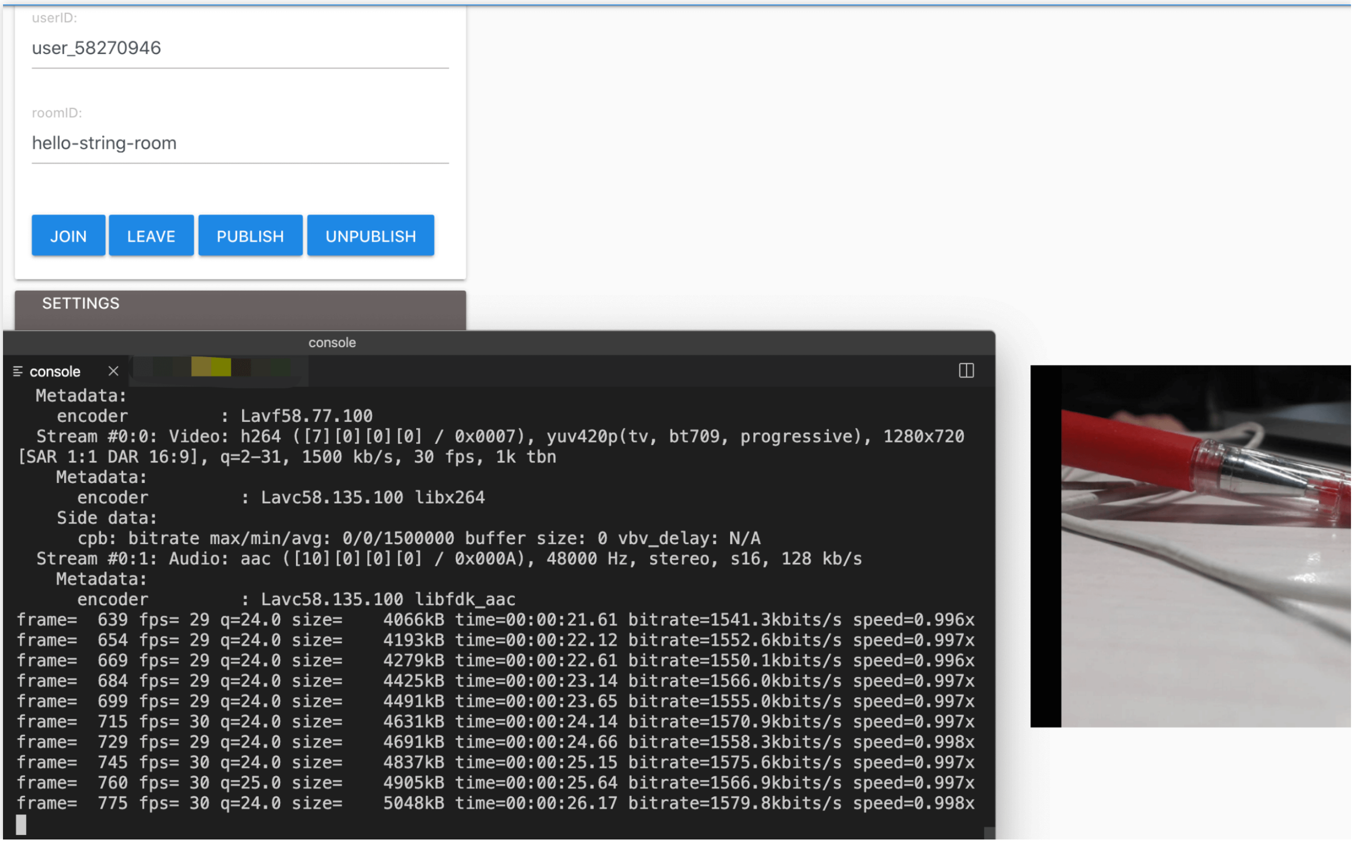
Option	Note
-c:v	The video encoding library. `libx264` is recommended.
-b:v	The video bitrate. For example, `1500k` means 1,500 Kbps.
-r	The video frame rate.
-profile:v	The video profile. If you set it to `baseline` , B-frames will not be encoded. The TRTC backend does not support B-frames.
-g	The GOP (keyframe interval).
-c:a	The audio encoding library. `libfdk_aac` is recommended.
-ac	The number of sound channels. Valid values: `1` , `2` .
-b:a	The audio bitrate.
-f	The container format. Set it to `flv` . The FLV container format is required to publish to TRTC.

Below is an example of reading a local file and publishing it to TRTC (note that quotation marks are required for the URL):

```
ffmpeg -loglevel debug -re -i sample.flv -c:v libx264 -preset ultrafast -profile:v
```

Playback on other clients

The screenshot below is an example of playback on the Web. You can also play the stream on other clients.


userID:
user_58270946
roomID:
hello-string-room
JOIN
LEAVE
PUBLISH
UNPUBLISH
SETTINGS
console
console
Metadata:
encoder : Lavf58.77.100
Stream #0:0: Video: h264 ([7][0][0][0] / 0x0007), yuv420p(tv, bt709, progressive), 1280x720 [SAR 1:1 DAR 16:9], q=2-31, 1500 kb/s, 30 fps, 1k tbn
Metadata:
encoder : Lavc58.135.100 libx264
Side data:
cpb: bitrate max/min/avg: 0/0/1500000 buffer size: 0 vbv_delay: N/A
Stream #0:1: Audio: aac ([10][0][0][0] / 0x000A), 48000 Hz, stereo, s16, 128 kb/s
Metadata:
encoder : Lavc58.135.100 libfdk_aac
frame= 639 fps= 29 q=24.0 size= 4066kB time=00:00:21.61 bitrate=1541.3kbits/s speed=0.996x
frame= 654 fps= 29 q=24.0 size= 4193kB time=00:00:22.12 bitrate=1552.6kbits/s speed=0.997x
frame= 669 fps= 29 q=24.0 size= 4279kB time=00:00:22.61 bitrate=1550.1kbits/s speed=0.996x
frame= 684 fps= 29 q=24.0 size= 4425kB time=00:00:23.14 bitrate=1566.0kbits/s speed=0.997x
frame= 699 fps= 29 q=24.0 size= 4491kB time=00:00:23.65 bitrate=1555.0kbits/s speed=0.997x
frame= 715 fps= 30 q=24.0 size= 4631kB time=00:00:24.14 bitrate=1570.9kbits/s speed=0.997x
frame= 729 fps= 29 q=24.0 size= 4691kB time=00:00:24.66 bitrate=1558.3kbits/s speed=0.998x
frame= 745 fps= 30 q=24.0 size= 4837kB time=00:00:25.15 bitrate=1575.6kbits/s speed=0.997x
frame= 760 fps= 30 q=25.0 size= 4905kB time=00:00:25.64 bitrate=1566.9kbits/s speed=0.997x
frame= 775 fps= 30 q=24.0 size= 5048kB time=00:00:26.17 bitrate=1579.8kbits/s speed=0.998x

Utilizing Beautification Effects

Web

Last updated：2024-04-16 16:35:01

Function Description

This document primarily describes how to use the beauty plugin.

Prerequisites

To use the Virtual Background Feature with SDKAppID, please make sure you have subscribed to the **RTC-Engine Professional Edition Monthly Package**. For related instructions on the Monthly Package, please refer to the Billing overview in the RTC-Engine documentation.

TRTC Web SDK version must be ≥ 5.5.0.

The requirements for different systems and configurations on the Web platform are as shown in the table below:

Platform	Operating System	Browser Version	FPS	Recommended Configuration	Remarks
Web	Windows	Chrome 90+ Firefox 90+ Edge 97+	30	Memory: 16GB CPU: i5-10500 GPU: Dedicated 2GB	It is recommended to use the latest version of Chrome browser (Enable hardware acceleration in the browser)
			15	Memory: 8GB CPU: i3-8300 GPU: Integrated Intel 1GB	
	Mac	Chrome 98+ Firefox 96+ Safari 14+	30	2019 MacBookMemory: 16GB (2667MHz) CPU: i7 (6-core 2.60GHz) GPU: AMD Radeon 5300M	
	Android	Chrome Firefox	30	High-end devices (e.g., Qualcomm Snapdragon 8 Gen1)	It is recommended to use Chrome, or Firefox browsers and other
			20	Mid-range devices (e.g., MediaTek Dimensity 8000-MAX)	
			10	Low-end devices (e.g., Qualcomm Snapdragon 660)	

					mainstream browsers
	iOS	Chrome Safari Firefox	30	iPhone 13	Requires iOS 14.4 or above- It is recommended to use Chrome or Safari browsers
			20	iPhone XR	

Implementation Steps

Step 1: Register the Plugin

Before using it, you need to register the plugin. Below is an example code for registering the plugin:

```
import TRTC from 'trtc-sdk-v5';
import { Beauty } from 'trtc-sdk-v5/plugins/video-effect/beauty';

const trtc = TRTC.create({ plugins: [Beauty] });
```

Step 2: Start the Local Camera

```
await trtc.startLocalVideo({
  view: 'local-video' // Fill in the div id
});
```

Step 3: Using Beauty and built-in Effects

Beauty Effect

The beauty effect can be controlled using three interfaces for starting, updating, and stopping:

`trtc.startPlugin('Beauty', options)` to enable the beauty effect.

`trtc.updatePlugin('Beauty', options)` to update the beauty effect parameters.

`trtc.stopPlugin('Beauty')` to stop the beauty effect.

The `options` parameter supports six beauty effect parameters:

```
type BeautifyOptions = {
  whiten?: number; // Whitening 0-1
  dermabrasion?: number; // Dermabrasion 0-1
  lift?: number; // Face Slimming 0-1
  shave?: number; // Jaw Shaving 0-1
  eye?: number; // Enlarged Eyes 0-1
  chin?: number; // Chin Adjustment 0-1
```

```
}
```

Built-in Effects

The plugin offers build-in video effects, here is the list:

EffectId	Name
EDE5D18065451445	Milk-Bottle-Face-Mask
7A62218064EF7959	Felt-Hairband
B4D63180653948C3	Bangs-Headband
D7C6418064B90F4C	Cute-Kitty
1D6EB18069D76A62	Polka-Dot-Girl
CBE7618065D9D76F	Heart-Fireworks
9B86618065596018	Cute-Piggy
428C518065369ACF	Double-Braids
B30E218064F7B397	Vibrant-Stickers
AE3C81806521B49B	Rabbit-Sauce

We use `EffectId` to identify the effect which will be apply.


```
type EffectListOptions = {
  id: string;
  intensity?: number; // specify the strength of the effect
}
```

Example Code

When starting the plugin, the `options` should also include the current user's login information (sdkAppId, userId, userSig) for verification.

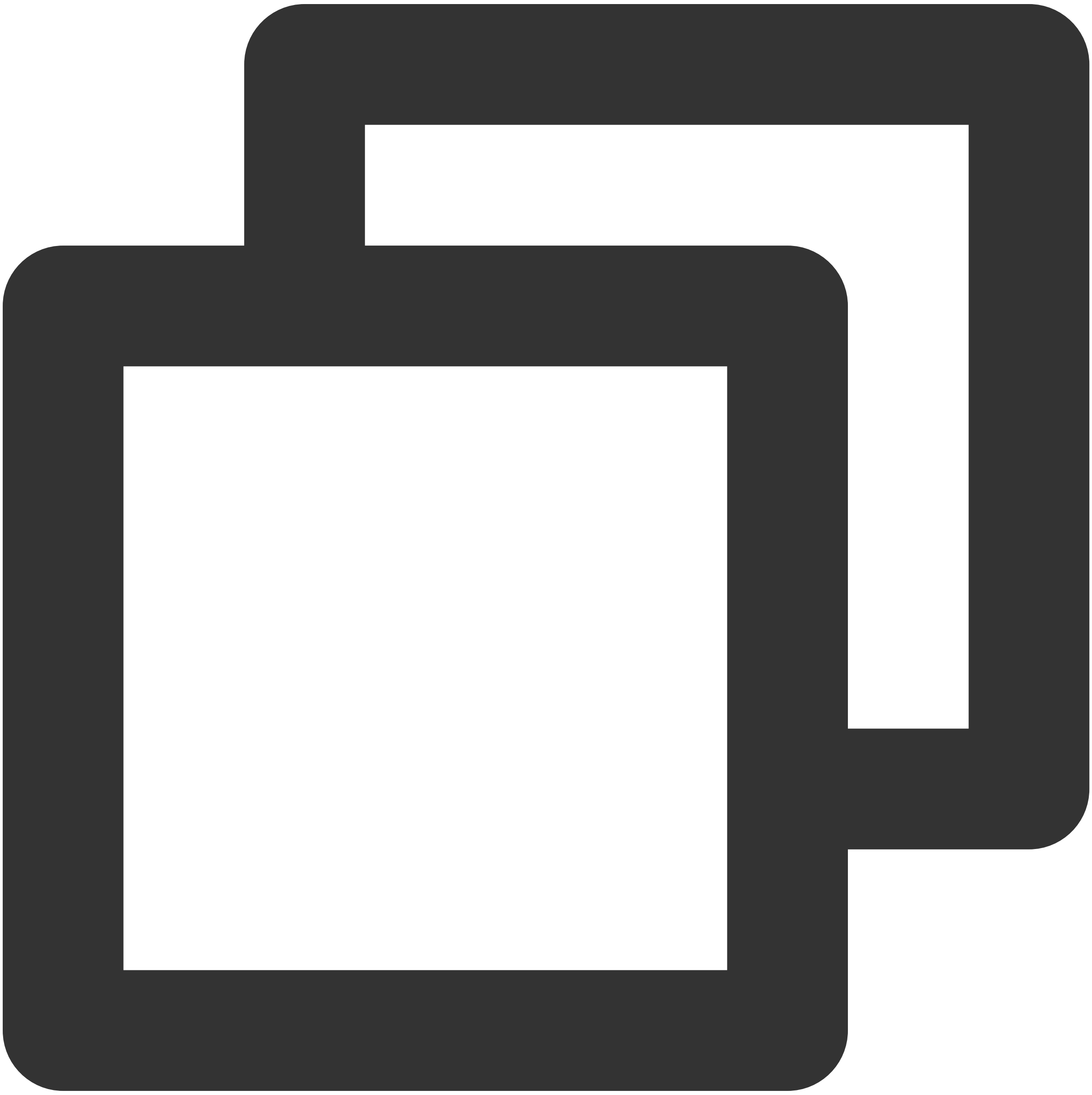
Here are some code examples:


```
// Enable the plugin
const options = {
  sdkAppId: 0,
  userId: '',
  userSig: '',

  // BeautifyOptions
  whiten: 0.5, // Whitening 0-1
  dermabrasion: 0.5, // Dermabrasion 0-1
  lift: 0.5, // Face Slimming 0-1
  shave: 0.5, // Jaw Shaving 0-1
```

```
    eye: 0.5, // Enlarged Eyes 0-1
    chin: 0.5, // Chin Adjustment 0-1

    // EffectListOptions Array
    effect: [{
      id: '7A62218064EF7959', // the effect id
      intensity: 0.7 // specify the strength of the effect
    }]
  }
  try {
    await trtc.startPlugin('Beauty', options);
  } catch (error) {
    console.error('Beauty start failed', error);
  }
```



```
// Update beauty parameters
const options = {
  whiten: 0.5, // Whitening 0-1
  dermabrasion: 0.5, // Dermabrasion 0-1
  lift: 0.5, // Face Slimming 0-1
  shave: 0.5, // Jaw Shaving 0-1
  eye: 0.5, // Enlarged Eyes 0-1
  chin: 0.5, // Chin Adjustment 0-1

  effect: [{
    id: '7A62218064EF7959', // the effect id
```

```
    intensity: 0.7, // specify the strength of the effect
  },{
    id: 'D7C6418064B90F4C', // the effect id
    intensity: 0.7, // specify the strength of the effect
  }]
}
try {
  await trtc.updatePlugin('Beauty', options);
} catch (error) {
  console.error('Beauty update failed', error);
}
```

```
// Stop the beauty effect
try {
  await trtc.stopPlugin('Beauty');
} catch (error) {
  console.error('Beauty stop failed', error);
}
```

API Documentation

trtc.startPlugin('Beauty', options)

Enable the beauty effect.

options

Name	Type	Attributes	Description
sdkAppId	`number`	Required	Current application ID
userId	`string`	Required	Current user ID
userSig	`string`	Required	UserSig corresponding to the user ID
whiten	`number`	Optional	Whitening, range [0,1]
dermabrasion	`number`	Optional	Dermabrasion, range [0,1]
lift	`number`	Optional	Face slimming, range [0,1]
shave	`number`	Optional	Jaw shaving, range [0,1]
eye	`number`	Optional	Enlarged eyes, range [0,1]
chin	`number`	Optional	Chin adjustment, range [0,1]
effect	`Array<EffectListOptions>`	Optional	effect list
onError	`(error) => {}`	Optional	Callback for errors that occur during runtime error.code=10000003 indicates high rendering latency error.code=10000006 indicates insufficient browser feature support, which may lead to lagging.

			Recommended solutions can be found in the Common Issues section at the end of the document.

EffectListOptions:

Name	Type	Attributes	Description
id	`number`	Required	The effect id
intensity	`string`	Required	the strength of the effect

trtc.updatePlugin('Beauty', options)

Allows modification of beauty parameters.

options

Name	Type	Attributes	Description
whiten	`number`	Optional	Whitening, range [0,1]
dermabrasion	`number`	Optional	Dermabrasion, range [0,1]
lift	`number`	Optional	Face slimming, range [0,1]
shave	`number`	Optional	Jaw shaving, range [0,1]
eye	`number`	Optional	Enlarged eyes, range [0,1]
chin	`number`	Optional	Chin adjustment, range [0,1]
effect	`Array<EffectListOptions>`	Optional	effect list

EffectListOptions:

Name	Type	Attributes	Description
id	`number`	Required	The effect id
intensity	`string`	Required	the strength of the effect

trtc.stopPlugin('Beauty')

Disables the beauty effect.

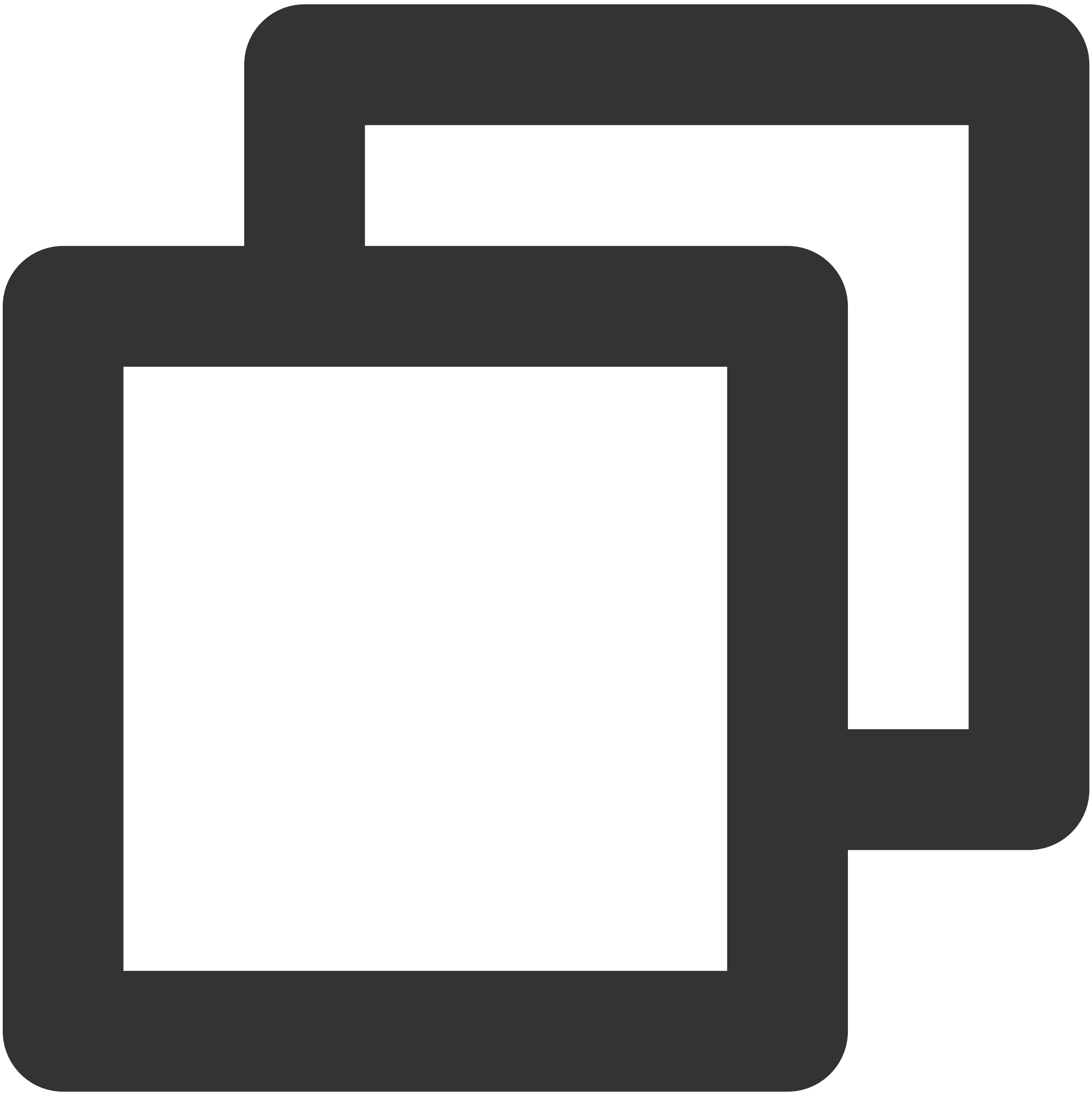
Frequently Asked Questions

1. **When running the demo in Chrome, the video appears upside down and laggy**

This plugin uses GPU acceleration, and you need to enable hardware acceleration mode in your browser settings. You can copy and paste `chrome://settings/system` into your browser's address bar and enable hardware acceleration mode.

2. **When device performance is insufficient and causes high latency with long rendering times**

You can reduce video resolution or frame rate by listening to events.




```
async function onError(error) {
  const { code } = error;
  if (code === 10000003 || code === 10000006) {
    // Reduce resolution and frame rate
    await trtc.updateLocalVideo({
      option: {
        profile: '480p_2'
      },
    });
    // await trtc.stopPlugin('Beauty'); // Or disable the plugin
  }
}

await trtc.startPlugin('Beauty', {
  ...// Other parameters
  onError,
});
```

Testing Hardware Devices Android&iOS&Windows&Mac

Last updated：2023-09-28 11:53:14

Overview

Given that it is difficult for users to detect device problems during a call, we recommend that you test devices such as cameras and mics before a video call.

Supported Platforms

iOS	Android	macOS	Windows	Electron	Web
×	×	✓	✓	✓	✓ (Web)

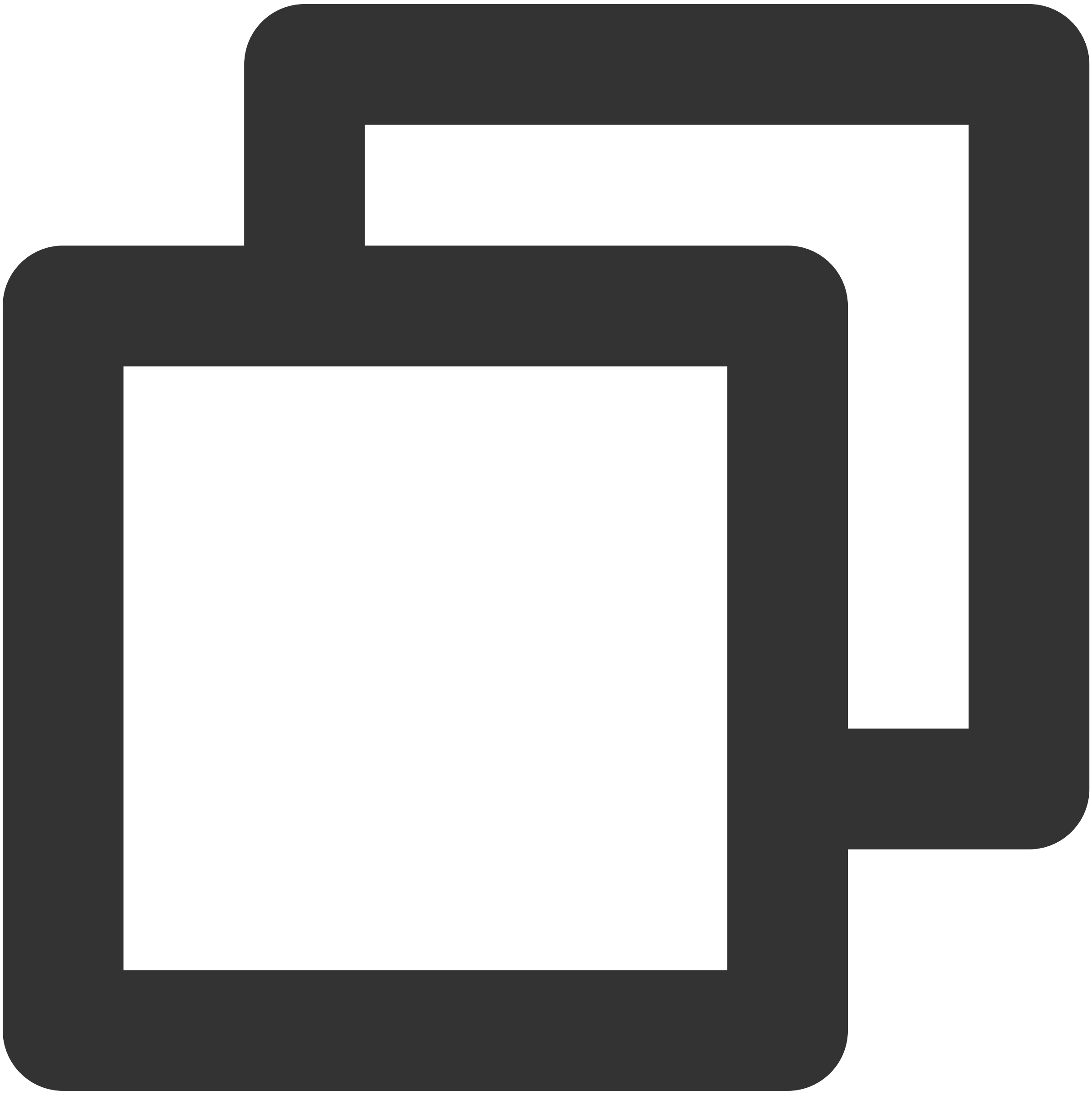
Testing Camera

You can use the `startCameraDeviceTestInView` API of `TRTCCloud` to test a camera and can use the `setCurrentCameraDevice` API to switch cameras during testing.

macOS

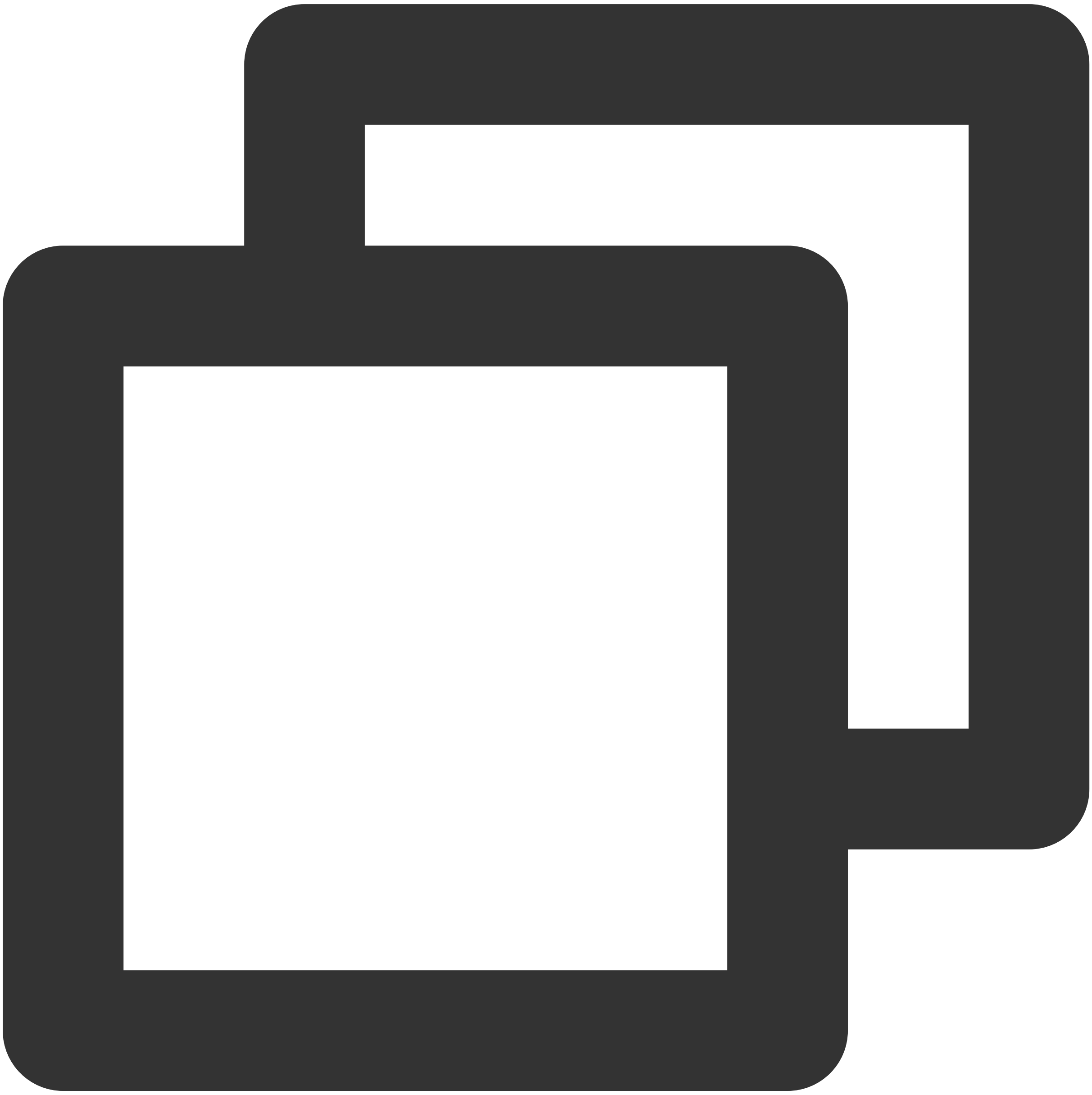
Windows (C++)

Windows (C#)



```
// Open the camera testing page, on which you can preview camera images and switch
- (IBAction)startCameraTest:(id)sender {
    // Start camera testing. `cameraPreview` is `NSView` on macOS or `UIView` on iO
    [self.trtcCloud startCameraDeviceTestInView:self.cameraPreview];
}

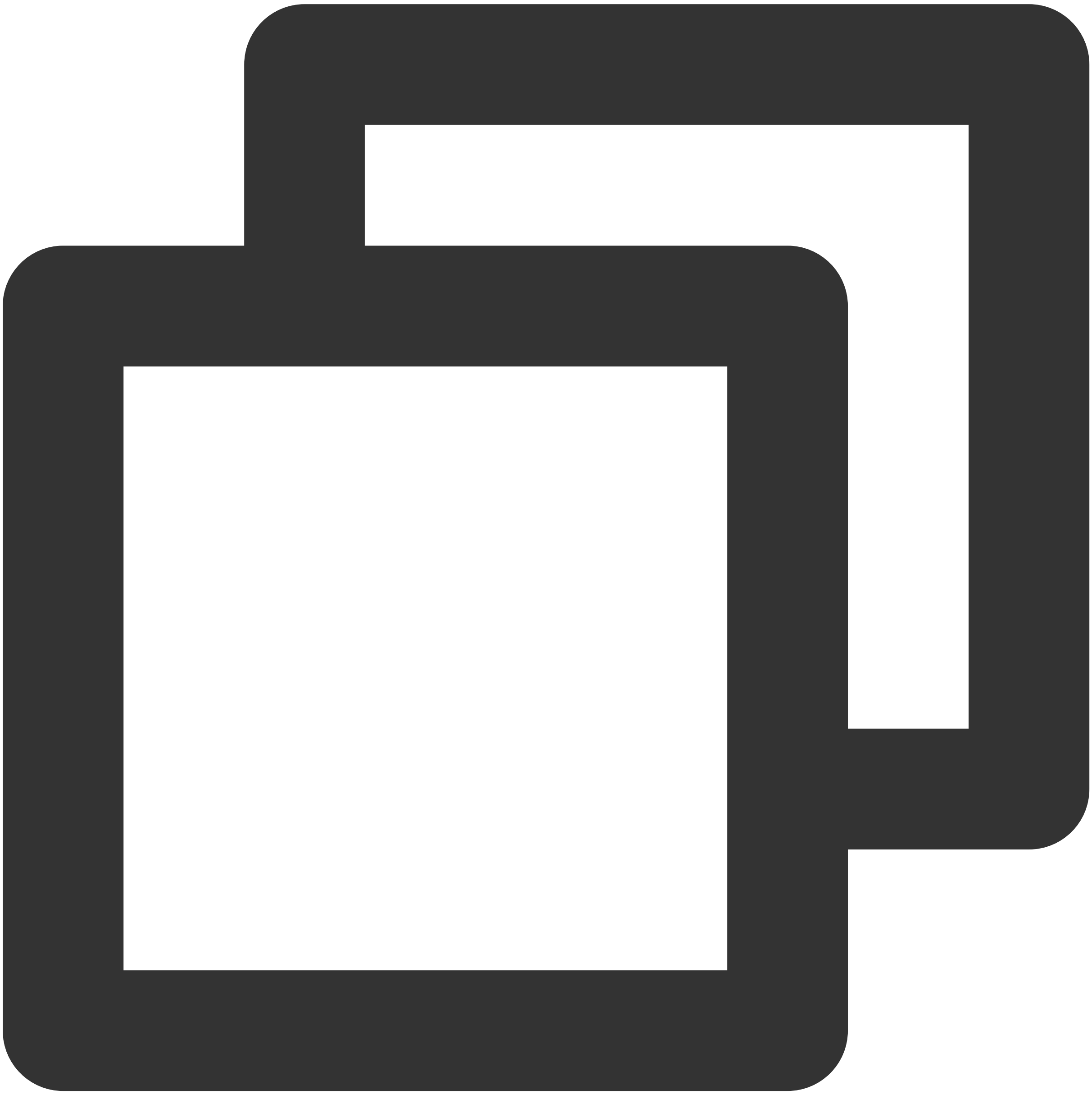
// Close the camera testing page.
- (void)windowWillClose:(NSNotification *)notification{
    // Stop camera testing.
    [self.trtcCloud stopCameraDeviceTest];
}
```



```
// Start camera testing. Pass in the control handle that renders video.
void TRTCMainViewController::startTestCameraDevice(HWND hwnd)
{
     trtcCloud->startCameraDeviceTest(hwnd);
}

// Stop camera testing.
void TRTCMainViewController::stopTestCameraDevice()
{
```

```
    trtcCloud->stopCameraDeviceTest();
}
```



```
// Start camera testing. Pass in the control handle that renders video.
private void startTestCameraDevice(Intptr hwnd)
{
    mTRTCCloud.startCameraDeviceTest(hwnd);
}

// Stop camera testing.
```

```
private void stopTestCameraDevice()
{
    mTRTCCloud.stopCameraDeviceTest();
}
```

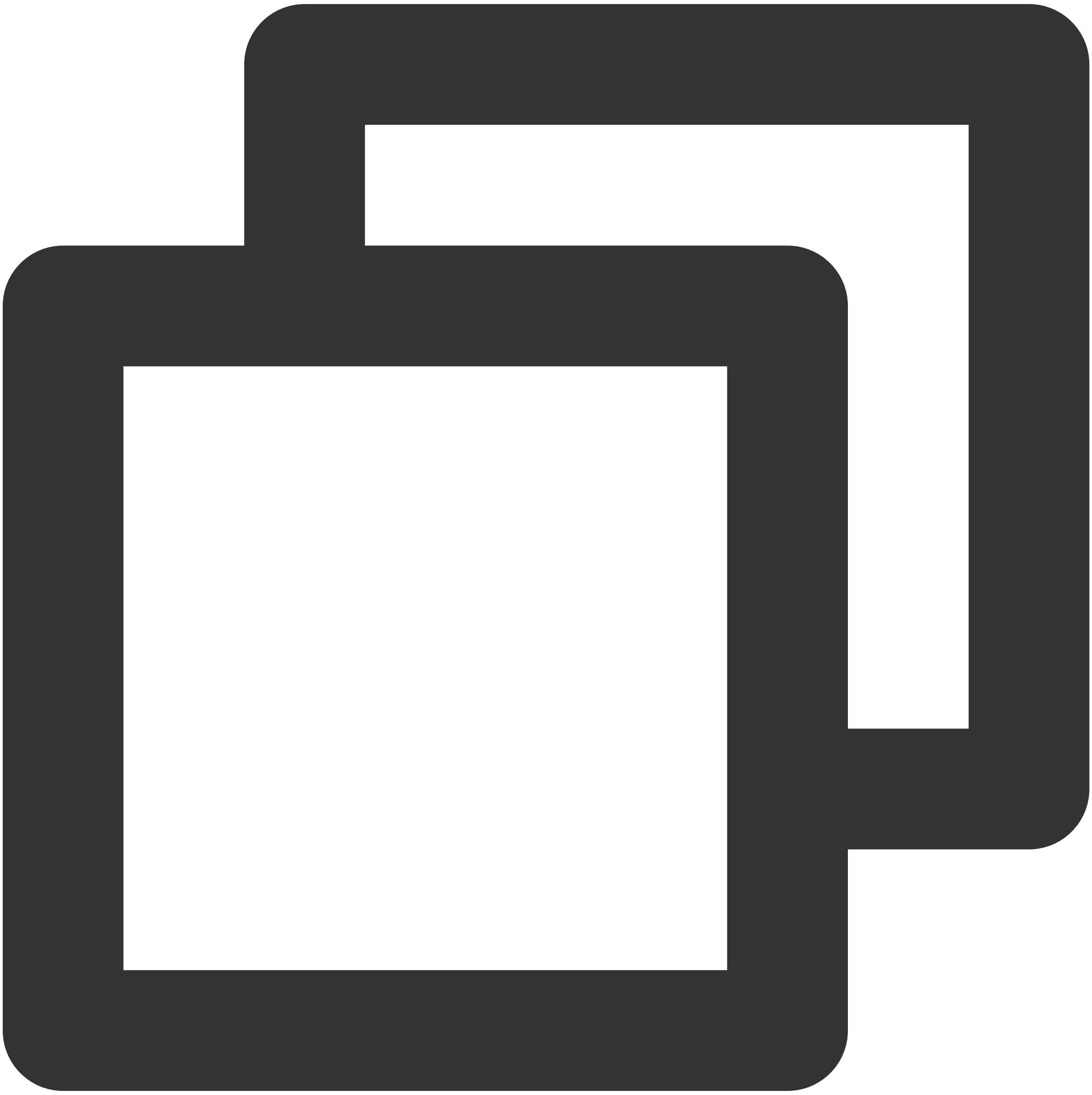
Testing Mic

You can use the `startMicDeviceTest` API of `TRTCCloud` to measure mic volume in real time. The result is returned via a callback.

macOS

Windows (C++)

Windows (C#)



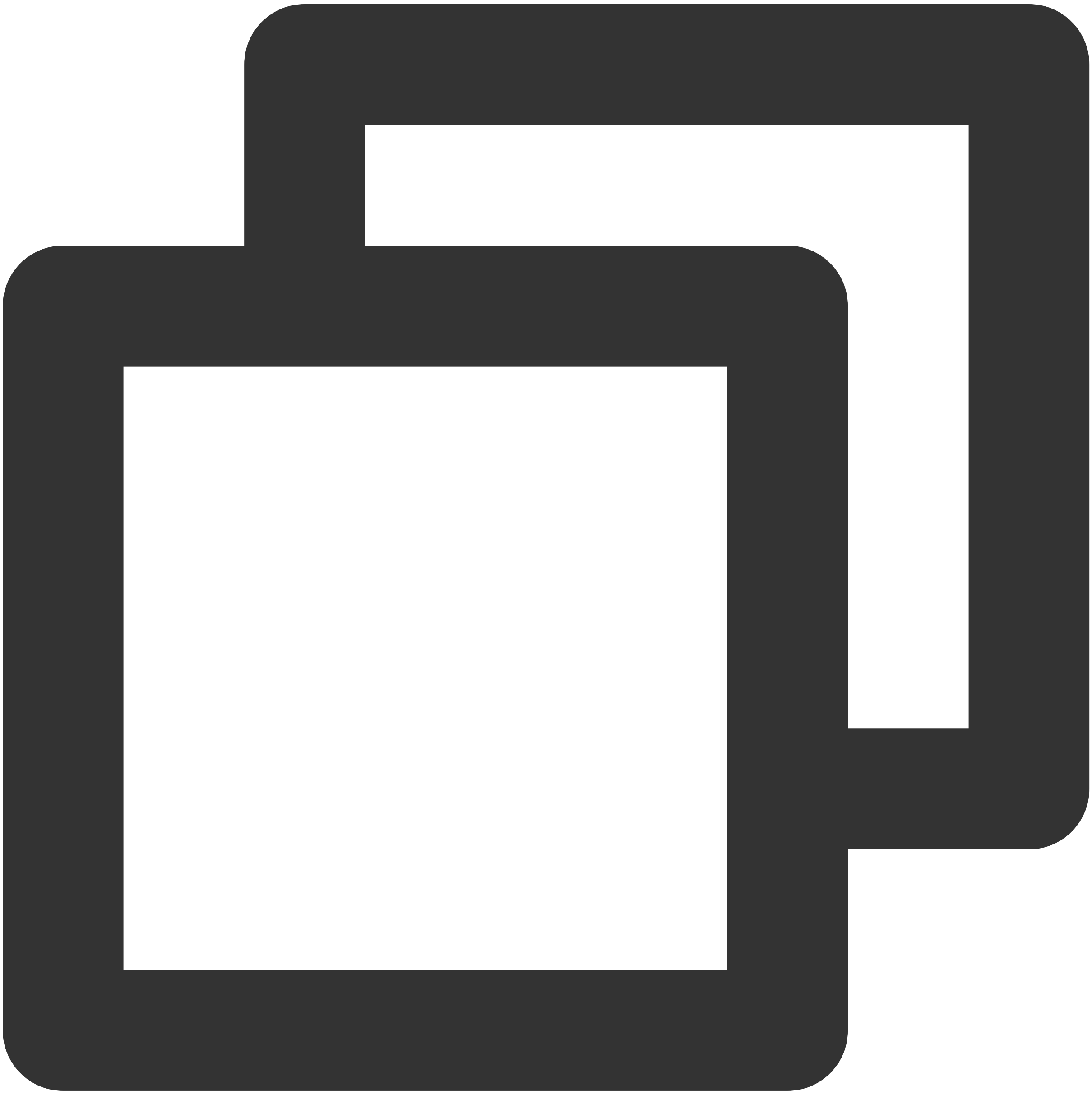
```
// Sample code for mic testing
-(IBAction)micTest:(id)sender {
  NSButton *btn = (NSButton *)sender;
  if (btn.state == 1) {
          // Start mic testing.
      __weak __typeof(self) wself = self;
      [self.trtcCloud startMicDeviceTest:500  testEcho:^(NSInteger volume) {
          dispatch_async(dispatch_get_main_queue(), ^{
                          // Refresh the mic volume bar.
              [wself _updateInputVolume:volume];
          });
```

```
        }];
        btn.title = @"Stop test";
    }
    else{
            // Stop mic testing.
        [self.trtcCloud stopMicDeviceTest];
        [self _updateInputVolume:0];
        btn.title = @"Start test";
    }
}
```



```
// Sample code for mic testing
void TRTCMainViewController::startTestMicDevice()
{
    // Set the interval for triggering the volume callback and listen for the `onTe
    uint32_t interval = 500;
    // Start mic testing.
    trtcCloud->startMicDeviceTest(interval);
}

// Stop mic testing.
void TRTCMainViewController::stopTestMicDevice()
{
     trtcCloud->stopMicDeviceTest();
}
```



```
// Sample code for mic testing
private void startTestMicDevice()
{
    // Set the interval for triggering the volume callback and listen for the `onTe
    uint interval = 500;
    // Start mic testing.
    mTRTCCloud.startMicDeviceTest(interval);
}

// Stop mic testing.
private void stopTestMicDevice()
```

```
{
      mTRTCCloud.stopMicDeviceTest();
}
```

Testing Speaker

You can use the `startSpeakerDeviceTest` API of `TRTCCloud` to test whether a speaker works properly by playing a default MP3 file.

macOS

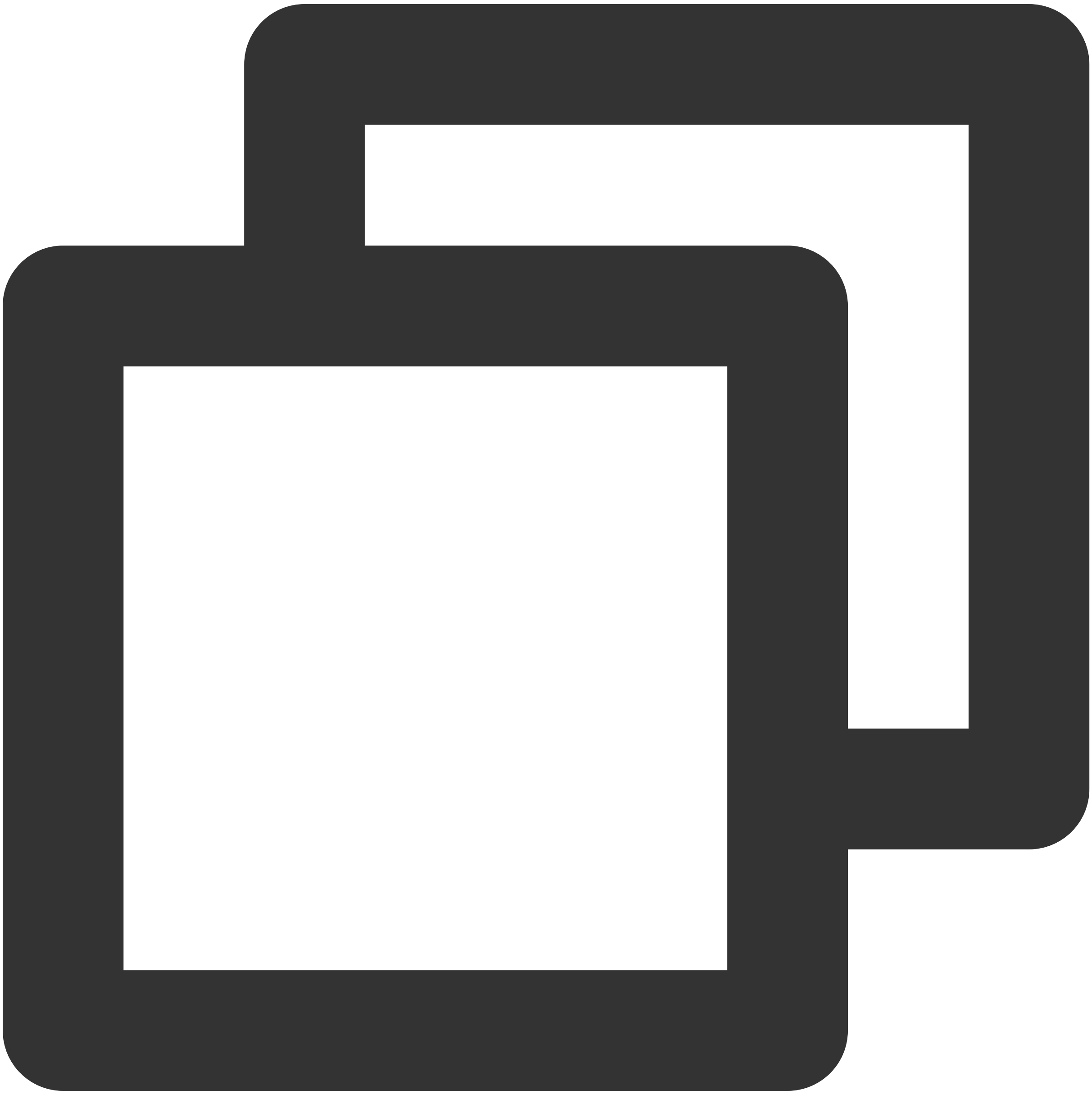
Windows (C++)

Windows (C#)


```
// Sample code for speaker testing
// Take an NSButton for example. In `xib`, set the title of the button in the on an
- (IBAction)speakerTest:(NSButton *)btn {
    NSString *path = [[NSBundle mainBundle] pathForResource:@"test-32000-mono" ofTy
    if (btn.state == NSControlStateValueOn) {
        // Click "Start Test".
        __weak __typeof(self) wself = self;
        [self.trtcEngine startSpeakerDeviceTest:path onVolumeChanged:^(NSInteger vo
            // The subsequent steps involve the UI and need to be executed in the m
            dispatch_async(dispatch_get_main_queue(), ^{
                // `_updateOutputVolume` means updating the speaker volume indicato
```

```
                [wself _updateOutputVolume:volume];
                if (playFinished) {
                    // Set the button status to "Start Test" after playback is comp
                    sender.state = NSControlStateValueOff;
                }
            });
        }];
    } else {
        // Click "Stop Test".
        [self.trtcEngine stopSpeakerDeviceTest];
        [self _updateOutputVolume:0];
    }
}

// Update the speaker volume indicator.
- (void)_updateOutputVolume:(NSInteger)volume {
    // `speakerVolumeMeter` is `NSLevelIndicator`.
    self.speakerVolumeMeter.doubleValue = volume / 255.0 * 10;
}
```

```
// Sample code for speaker testing
void TRTCMainViewController::startTestSpeakerDevice(std::string testAudioFilePath)
{
    // `testAudioFilePath` is the absolute path of the audio file (in WAV or MP3 fo
    // Listen for the `onTestSpeakerVolume` callback to get the speaker volume.
    trtcCloud->startSpeakerDeviceTest(testAudioFilePath.c_str());
}

// Stop speaker testing.
void TRTCMainViewController::stopTestSpeakerDevice() {
    trtcCloud->stopSpeakerDeviceTest();
```



```
}
```

```
// Sample code for speaker testing
private void startTestSpeakerDevice(string testAudioFilePath)
{
    // `testAudioFilePath` is the absolute path of the audio file (in WAV or MP3 fo
    // Listen for the `onTestSpeakerVolume` callback to get the speaker volume.
    mTRTCCloud.startSpeakerDeviceTest(testAudioFilePath);
}
```

```
// Stop speaker testing.
private void stopTestSpeakerDevice() {
    mTRTCCloud.stopSpeakerDeviceTest();
}
```

Web

Last updated：2023-11-16 15:09:42

Overview

Because it is difficult for users to detect device problems during a call, we recommend checking the browser and testing devices such as cameras and mics before starting a video call.

Browser Environment Check

Before enterRoom, we recommend you use the TRTC.isSupported API to check whether the SDK supports the current browser first, and if not, please recommend the user to use a supported browser, refer to Supported browsers.

```
TRTC.isSupported().then(checkResult => {
  // Not supported, guide the user to use a supported browser(Chrome 56+, Edge 80+,
  if (!checkResult.result) {}
  // Not support to publish video
  if (!checkResult.detail.isH264EncodeSupported) {}
  // Not support to subscribe video
  if (!checkResult.detail.isH264DecodeSupported) {}
})
```

If the check result returned by TRTC.isSupported is false, this may be because:

1. The web page uses the http protocol. Browsers do not allow http protocol sites to capture cameras and microphones, you need to deploy your page using the https protocol.
2. The current browser does not support WebRTC, you need to guide the user to use the recommended browser, refer to Supported browsers.
3. Firefox browser needs to load H264 codec dynamically after installation, so the detection result will be false for a short period of time, please wait and try again or guide to use other browsers.

Audio/Video Device Test

To ensure that users can have a good user experience with the TRTC SDK, we recommend you check the user's device and network conditions and provide troubleshooting suggestions before the user enters a TRTC room.
You can quickly integrate the device and network check features by referring to the following methods:
`rtc-detect` Library
React Component for Device Check
TRTC Capability Check Page

`rtc-detect` Library

You can use rtc-detect to check whether the current environment is supported by the TRTC SDK and view the details of the current environment.

Installation


```
npm install rtc-detect
```

Usage


```
import RTCDetect from 'rtc-detect';
// Initialize the detection module
const detect = new RTCDetect();
// Get the detection result of the current environment
const result = await detect.getReportAsync();
// `result` contains the current environment system information, API support, codec
console.log('result is: ' + result);
```

API

(async) isTRTCSupported()

This API is used to check whether the current environment supports TRTC.

```
const detect = new RTCDetect();
const data = await detect.isTRTCSupported();

if (data.result) {
  console.log('current browser supports TRTC.')
} else {
  console.log(`current browser does not support TRTC, reason: ${data.reason}.`)
}
```


getSystem()

This API is used to get the current system environment parameters.

Item	Type	Description
UA	string	The browser user-agent.
OS	string	The OS of the current device.
browser	object	The current browser information in the format of `{ name, version }`.
displayResolution	object	The current resolution in the format of `{ width, height }`.
getHardwareConcurrency	number	The number of CPU cores of the current device.

```
const detect = new RTCDetect();
const result = detect.getSystem();
```

getAPISupported()

This API is used to get the API support of the current environment.

Item	Type	Description
isUserMediaSupported	boolean	Whether the user media data stream can be obtained

isWebRTCSupported	boolean	Whether WebRTC is supported
isWebSocketSupported	boolean	Whether WebSocket is supported
isWebAudioSupported	boolean	Whether WebAudio is supported
isScreenCaptureAPISupported	boolean	Whether the screen stream can be obtained
isCanvasCapturingSupported	boolean	Whether the data stream can be obtained from the canvas
isVideoCapturingSupported	boolean	Whether the data stream can be obtained from the video
isRTPSenderReplaceTracksSupported	boolean	Whether renegotiation with `peerConnection` can be skipped when `track` is replaced
isApplyConstraintsSupported	boolean	Whether the camera resolution can be changed without calling `getUserMedia` again

```
const detect = new RTCDetect();
const result = detect.getAPISupported();
```

(async) getDevicesAsync()

This API is used to get the available devices in the current environment.

Item	Type	Description
hasWebCamPermissions	boolean	Whether the user camera data can be obtained

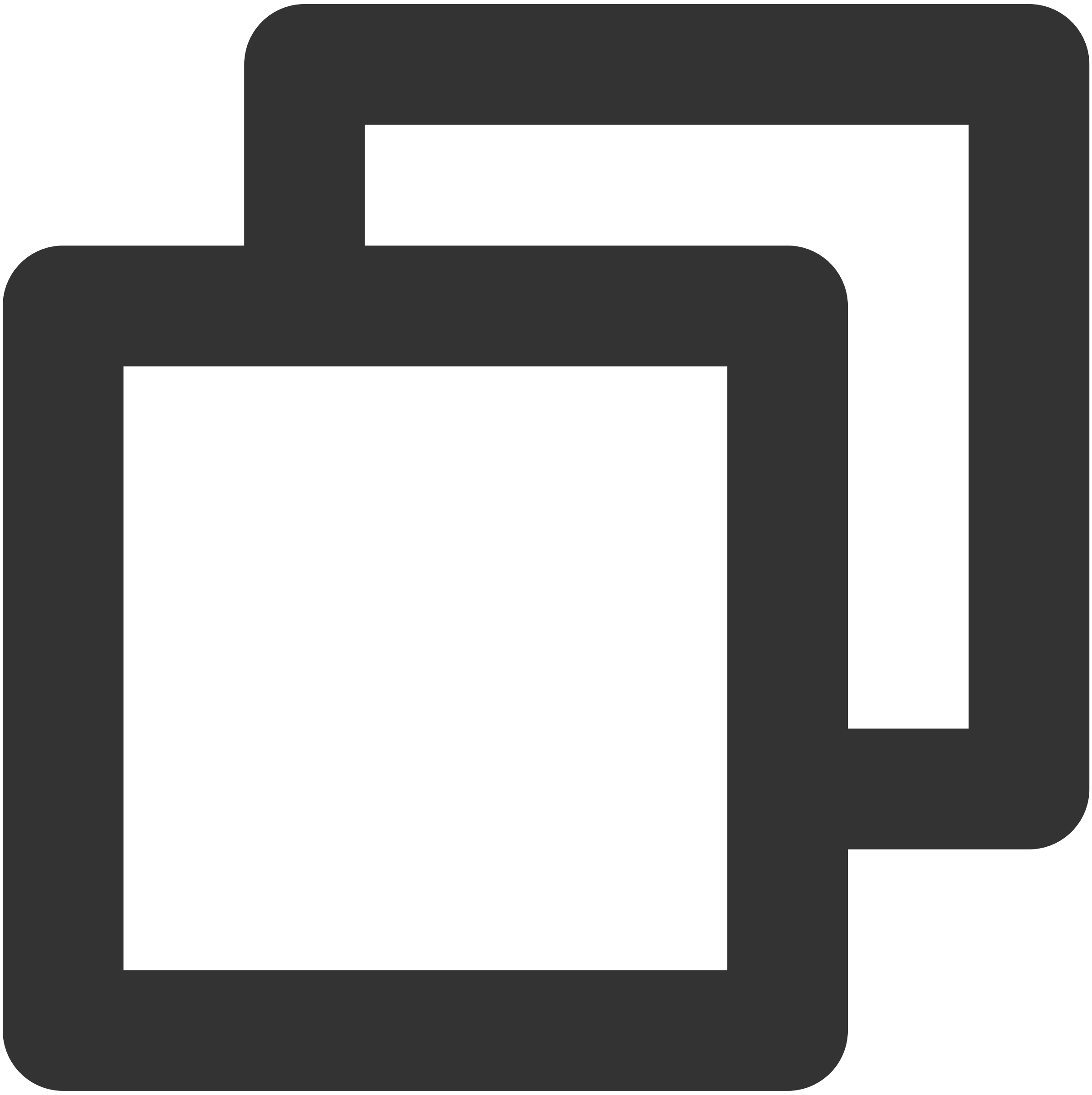
hasMicrophonePermission	boolean	Whether the user mic data can be obtained
cameras	array	List of user cameras, including their resolutions, maximum width, maximum height, and maximum frame rate (for certain browsers only) supported for video streams
microphones	array	List of mics used by users
speakers	array	List of speakers used by users

CameraItem

Item	Type	Description
deviceId	string	The device ID, which is usually unique and can be used to identify devices.
groupId	string	The group ID. If two devices belong to the same physical device, they have the same group ID.
kind	string	The camera device type: 'videoinput'.
label	string	A tag which describes the device.
resolution	object	The maximum resolution width, height, and frame rate supported by the camera {maxWidth: 1280, maxHeight: 720, maxFrameRate: 30}.

DeviceItem

Item	Type	Description
deviceId	string	The device ID, which is usually unique and can be used to identify devices.
groupId	string	The group ID. If two devices belong to the same physical device, they have the same group ID.
kind	string	The device type, such as 'audioinput' and 'audiooutput'.
label	string	A tag which describes the device.



```
const detect = new RTCDetect();
const result = await detect.getDevicesAsync();
```

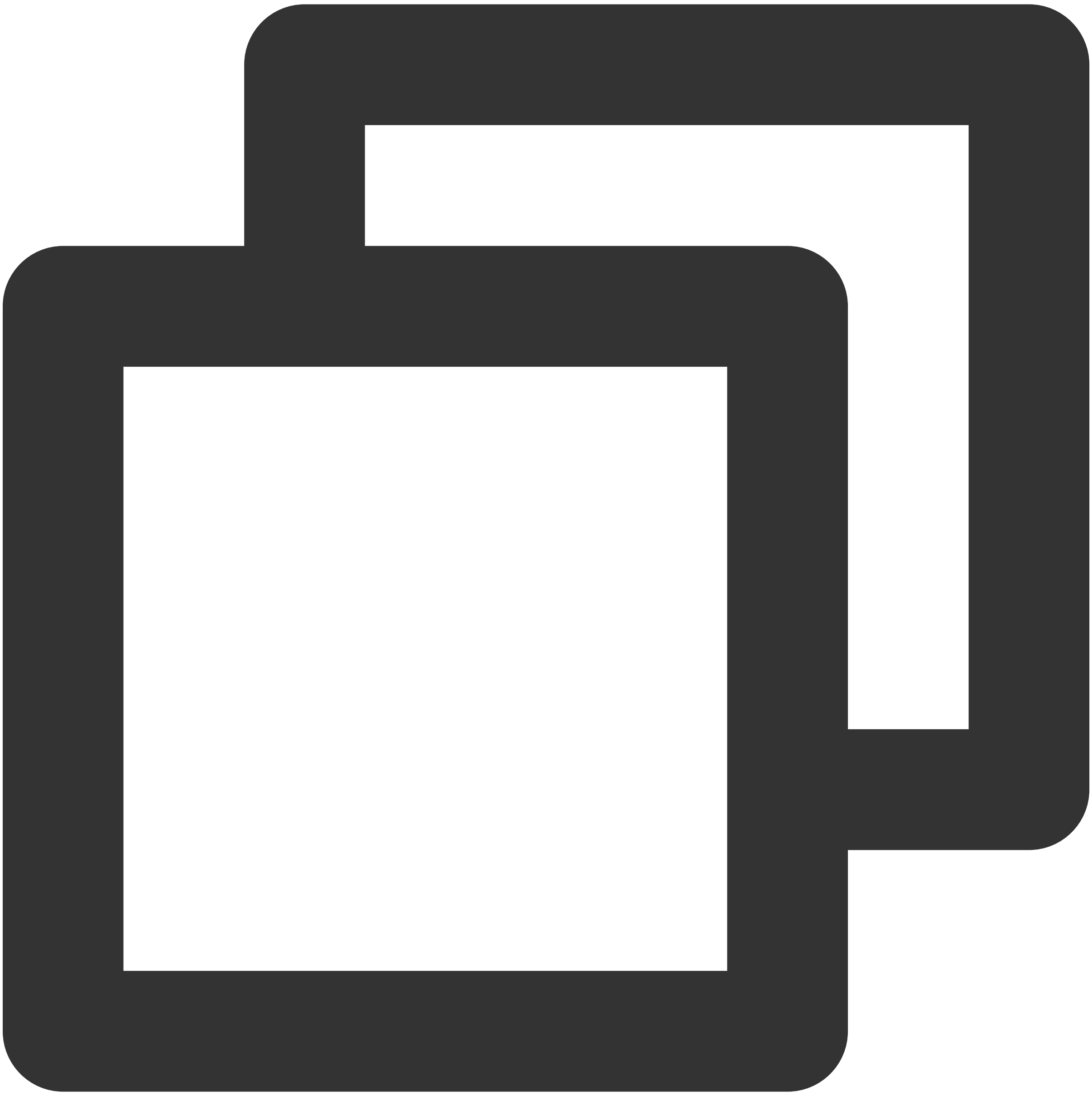
(async) getCodecAsync()

This API is used to get the codec support of the current environment.

Item	Type	Description
isH264EncodeSupported	boolean	Whether H.264 encoding is supported

isH264DecodeSupported	boolean	Whether H.264 decoding is supported
isVp8EncodeSupported	boolean	Whether VP8 encoding is supported
isVp8DecodeSupported	boolean	Whether VP8 decoding is supported

If encoding is supported, audio/video can be published. If decoding is supported, audio/video can be pulled for playback.




```
const detect = new RTCDetect();
const result = await detect.getCodecAsync();
```

(async) getReportAsync()

This API is used to get the detection report of the current environment.

Item	Type	Description
system	object	Same as the returned value of `getSystem()`
APISupported	object	Same as the returned value of `getAPISupported()`
codecsSupported	object	Same as the returned value of `getCodecAsync()`
devices	object	Same as the returned value of `getDevicesAsync()`

```
const detect = new RTCDetect();
const result = await detect.getReportAsync();
```

(async) isHardWareAccelerationEnabled()

This API is used to check whether hardware acceleration is enabled on the Chrome browser.

notice

The implementation of this API depends on the native WebRTC API. We recommend you call this API for check after calling `isTRTCSupported` . The check can take up to 30 seconds as tested below:

1. If hardware acceleration is enabled, this API will take about 2 seconds on Windows and 10 seconds on macOS.
2. If hardware acceleration is disabled, this API will take about 30 seconds on both Windows and macOS.

```
const detect = new RTCDetect();
const data = await detect.isTRTCSupported();

if (data.result) {
  const result = await detect.isHardWareAccelerationEnabled();
  console.log(`is hardware acceleration enabled: ${result}`);
} else {
```



```
  console.log(`current browser does not support TRTC, reason: ${data.reason}.`)
}
```

React Component for Device Check

Device check UI component features

1. Device connection and check logic processing
2. Network check logic processing
3. Optional network check tab
4. Support for Chinese and English

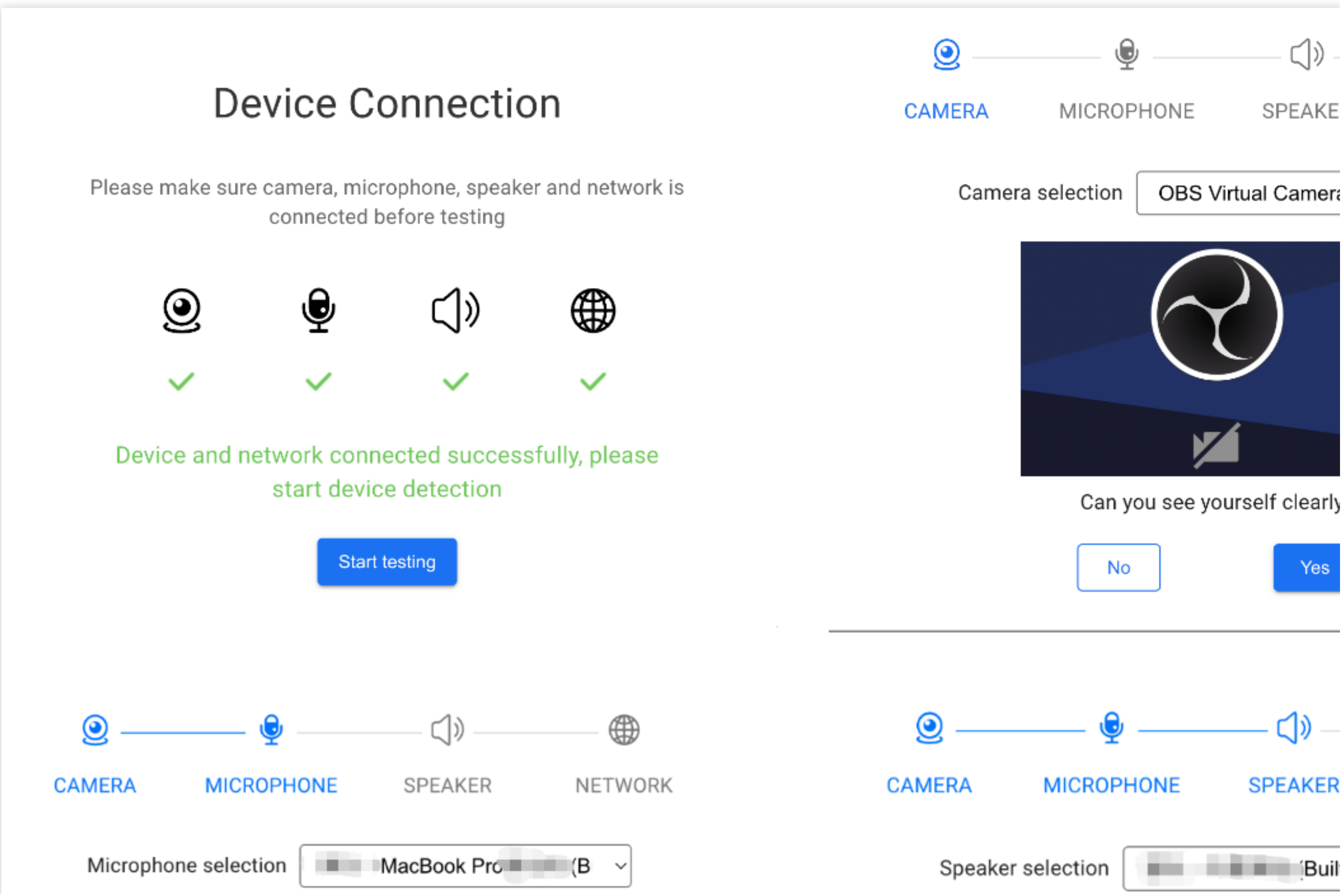
Device check UI component links

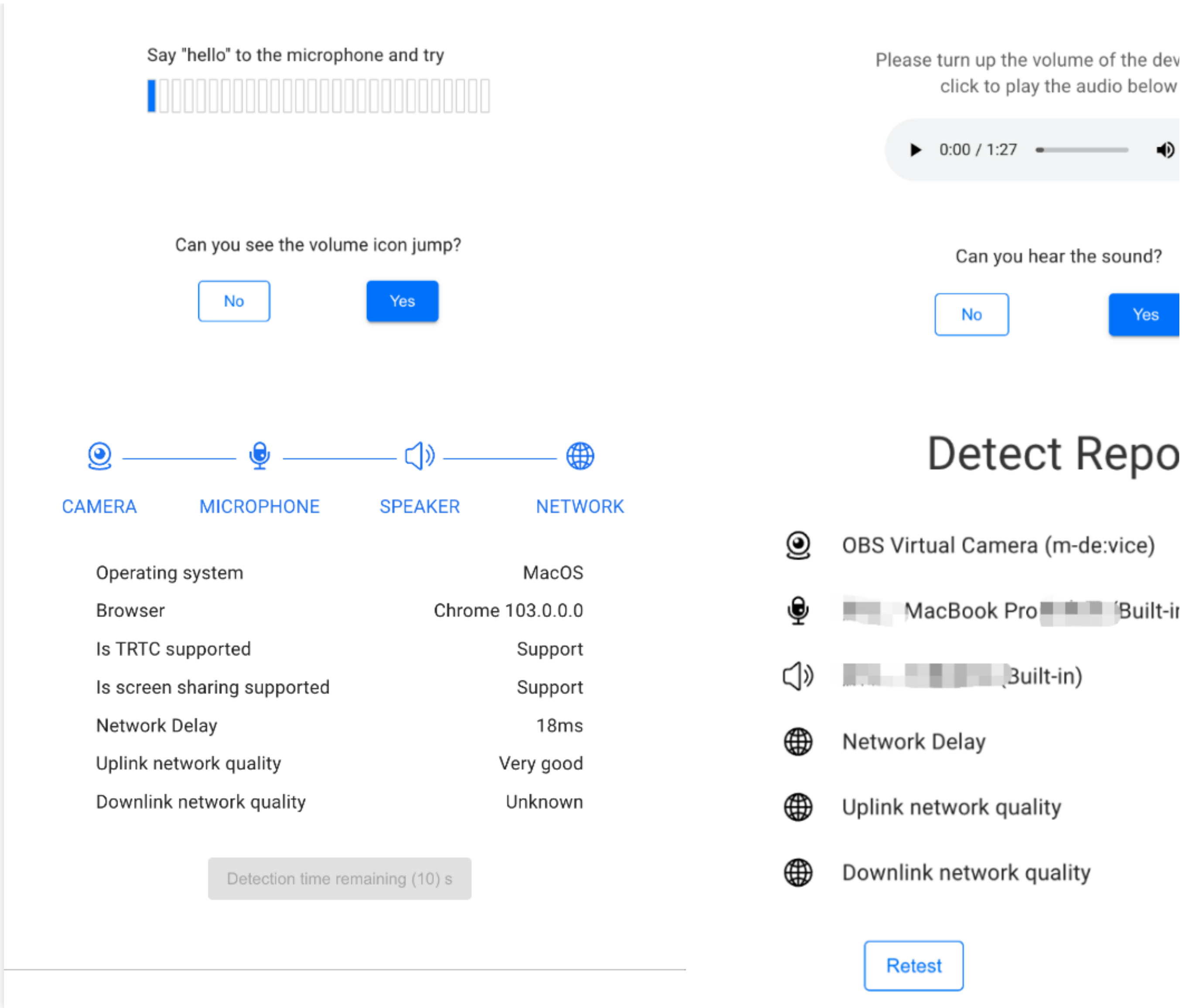
For more information on how to use the component's npm package, see rtc-device-detector-react.

For more information on how to debug the component's source code, see github/rtc-device-detector.

For more information on how to import the component, see WebRTC API Example.

Device check UI component page





Device and Network Detection Logic

1) Device Connection

The purpose of device connection is to detect whether the user's machine has camera, microphone, and speaker devices, and whether it is in a networked state. If there are camera and microphone devices, try to obtain audio and video streams and guide the user to grant access to the camera and microphone.

Determine whether the device has camera, microphone, and speaker devices

```
import TRTC from 'trtc-sdk-v5';

const cameraList = await TRTC.getCameraList();
const micList = await TRTC.getMicrophoneList();
const speakerList = await TRTC.getSpeakerList();
const hasCameraDevice = cameraList.length > 0;
const hasMicrophoneDevice = micList.length > 0;
const hasSpeakerDevice = speakerList.length > 0;
```

Obtain access to the camera and microphone


```
await trtc.startLocalVideo({ publish: false });
await trtc.startLocalAudio({ publish: false });
```

Check whether the device is connected to the network

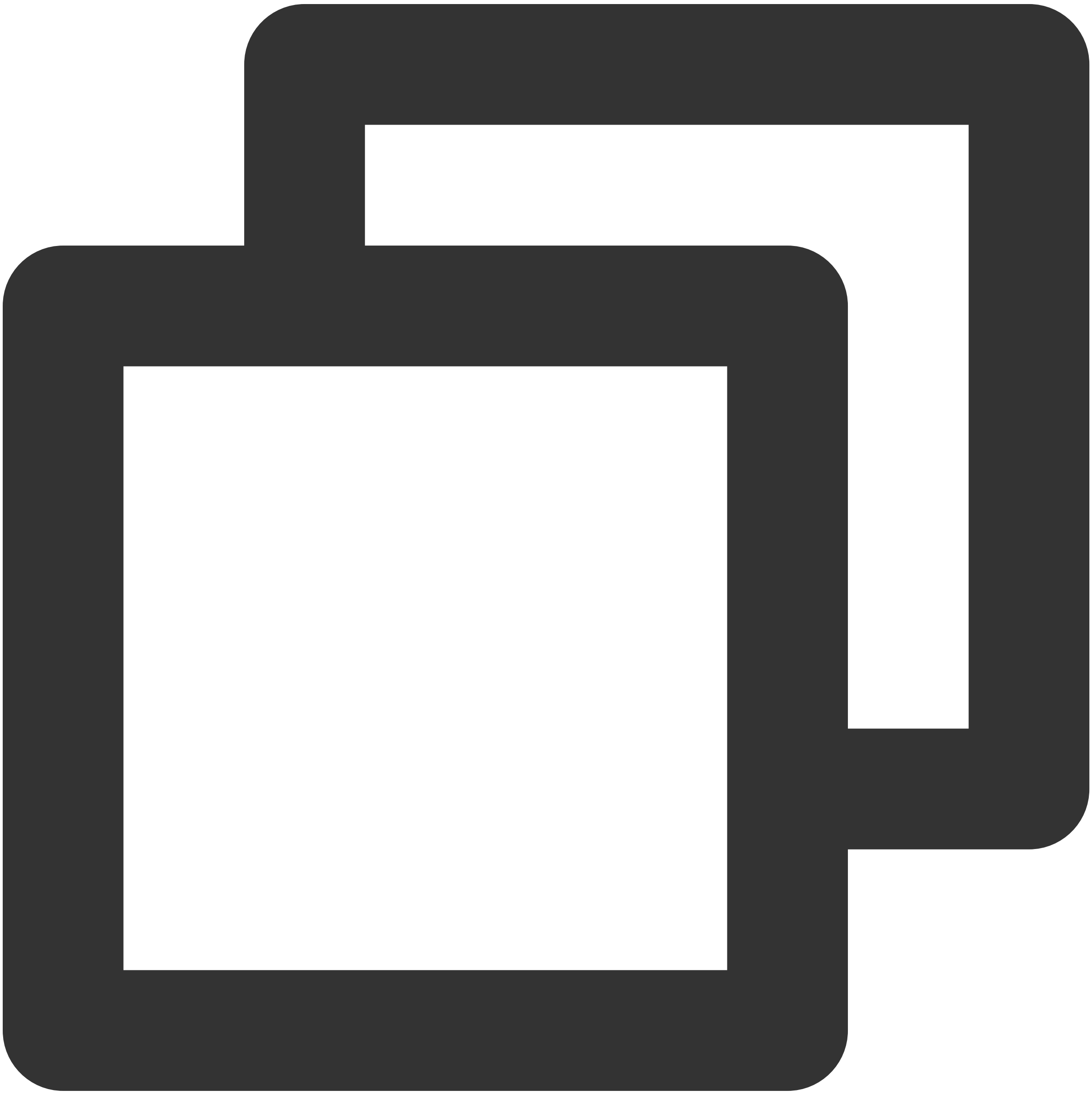
```
export function isOnline() {
  const url = 'https://web.sdk.qcloud.com/trtc/webrtc/assets/trtc-logo.png';
  return new Promise((resolve) => {
    try {
      const xhr = new XMLHttpRequest();
      xhr.onload = function () {
        resolve(true);
      };
      xhr.onerror = function () {
        resolve(false);
      };
```

```
        xhr.open('GET', url, true);
        xhr.send();
      } catch (err) {
        // console.log(err);
      }
    });
  }
  const isOnline = await isOnline();
```

2） Camera Detection

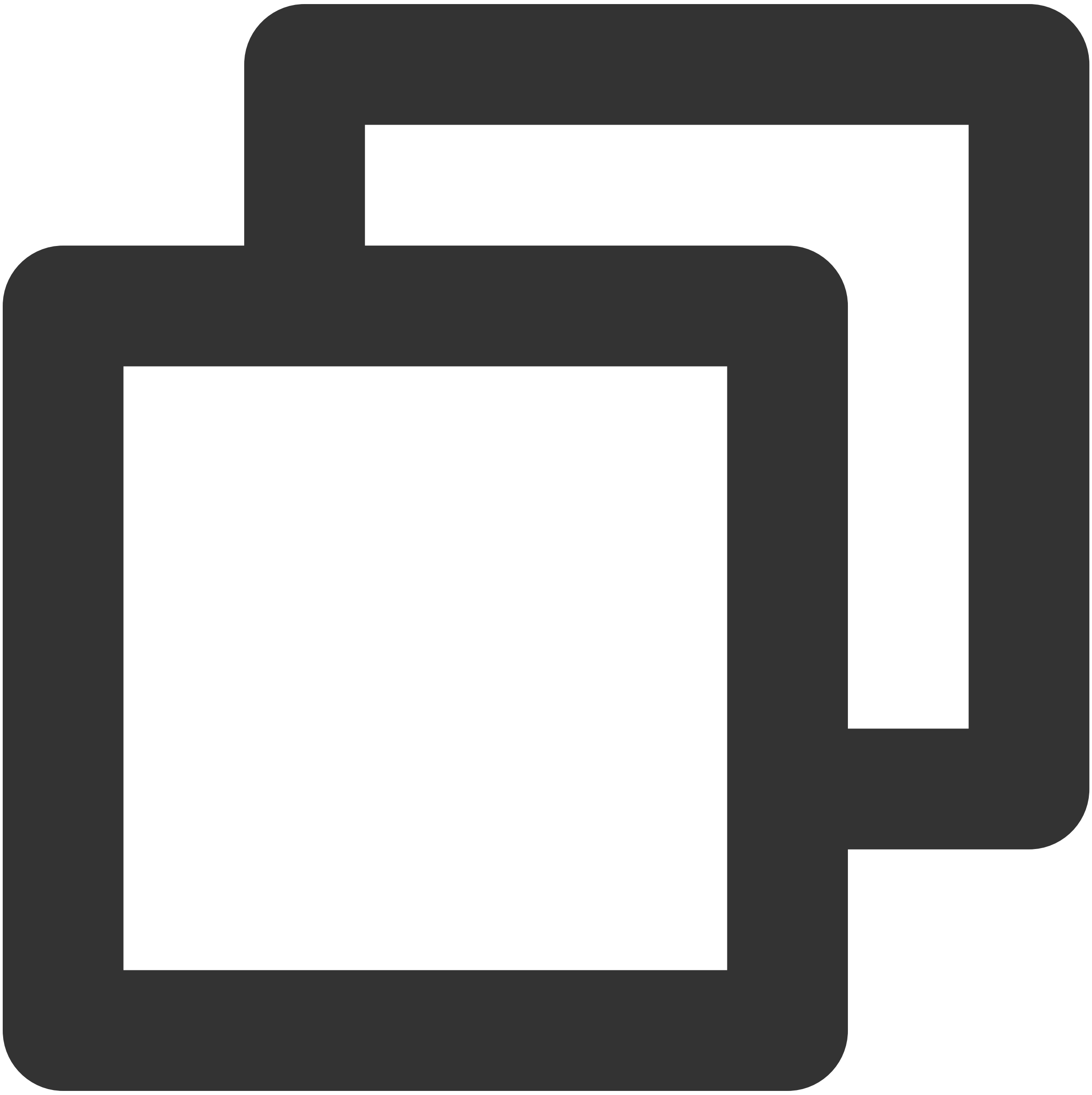
Detection principle: Open the camera and render the camera image on the page.

Open the camera



```
trtc.startLocalVideo({ view: 'camera-video', publish: false });
```

Switch camera



```
trtc.updateLocalVideo({
  option: { cameraId }
});
```

Device plug and unplug detection

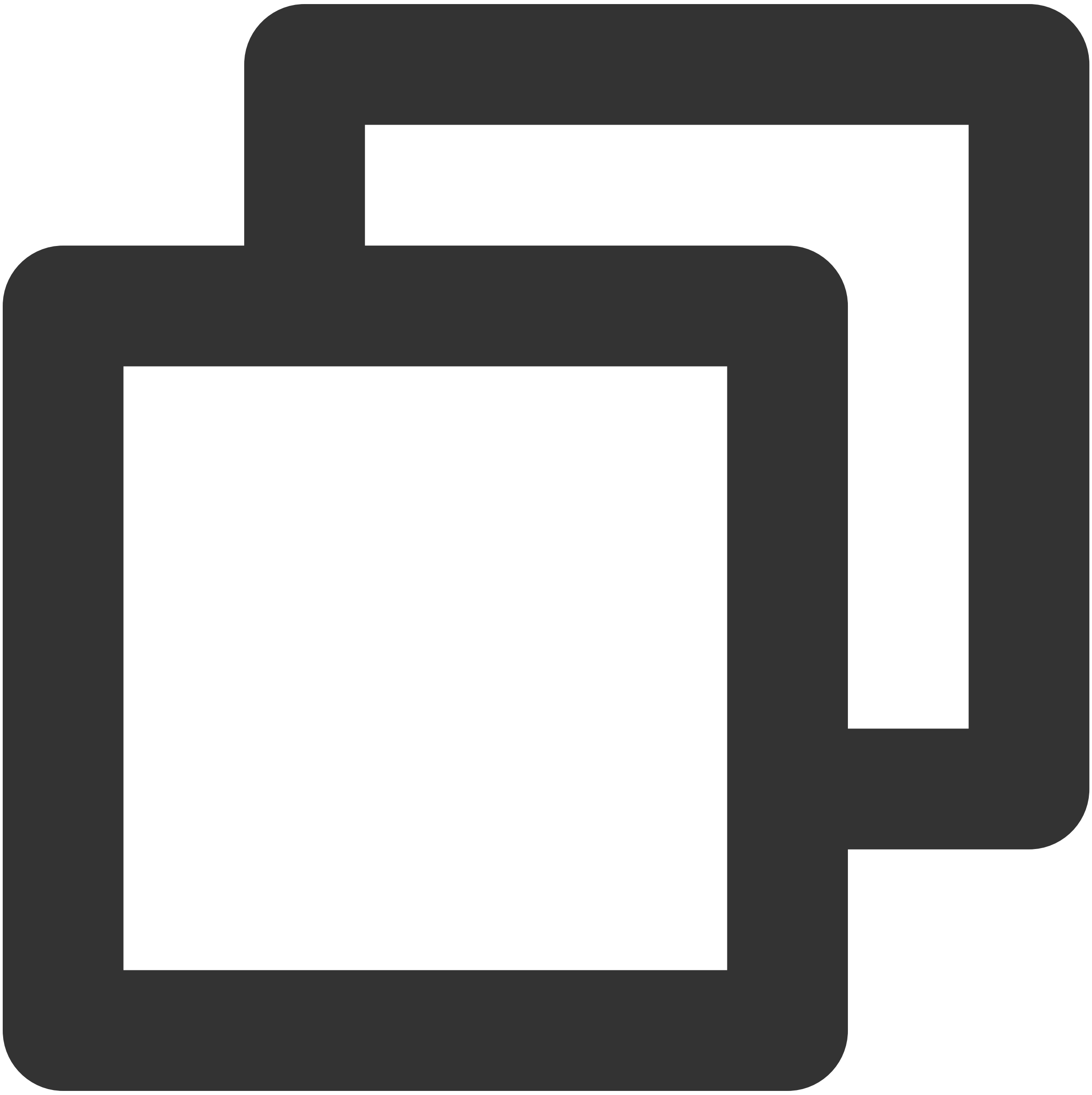
Close the camera after the detection is complete


```
trtc.stopLocalVideo();
```

3) Microphone Detection

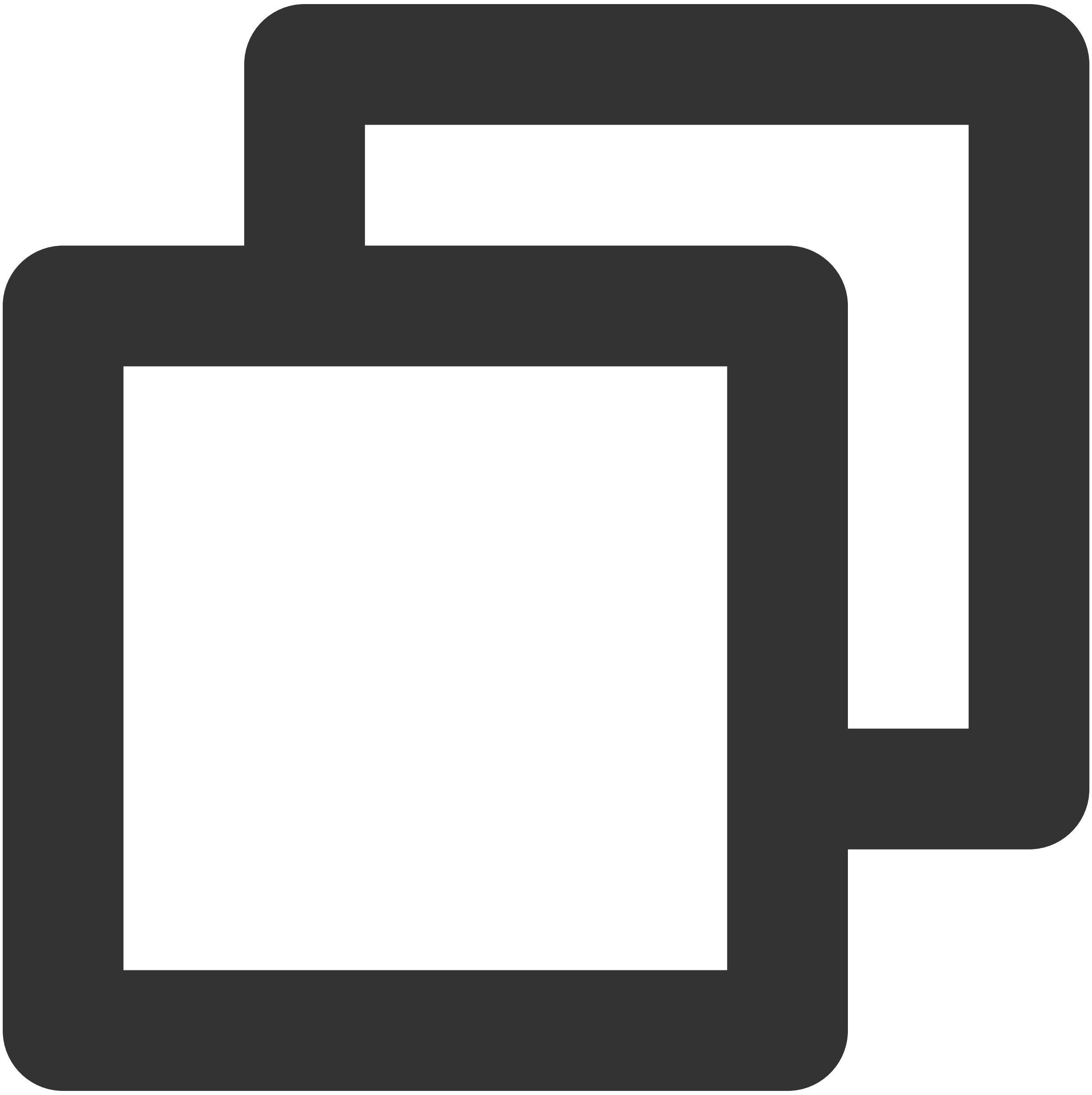
Detection principle: Open the microphone and obtain the microphone volume.

Open the microphone



```
trtc.startLocalAudio({ publish: false });
```

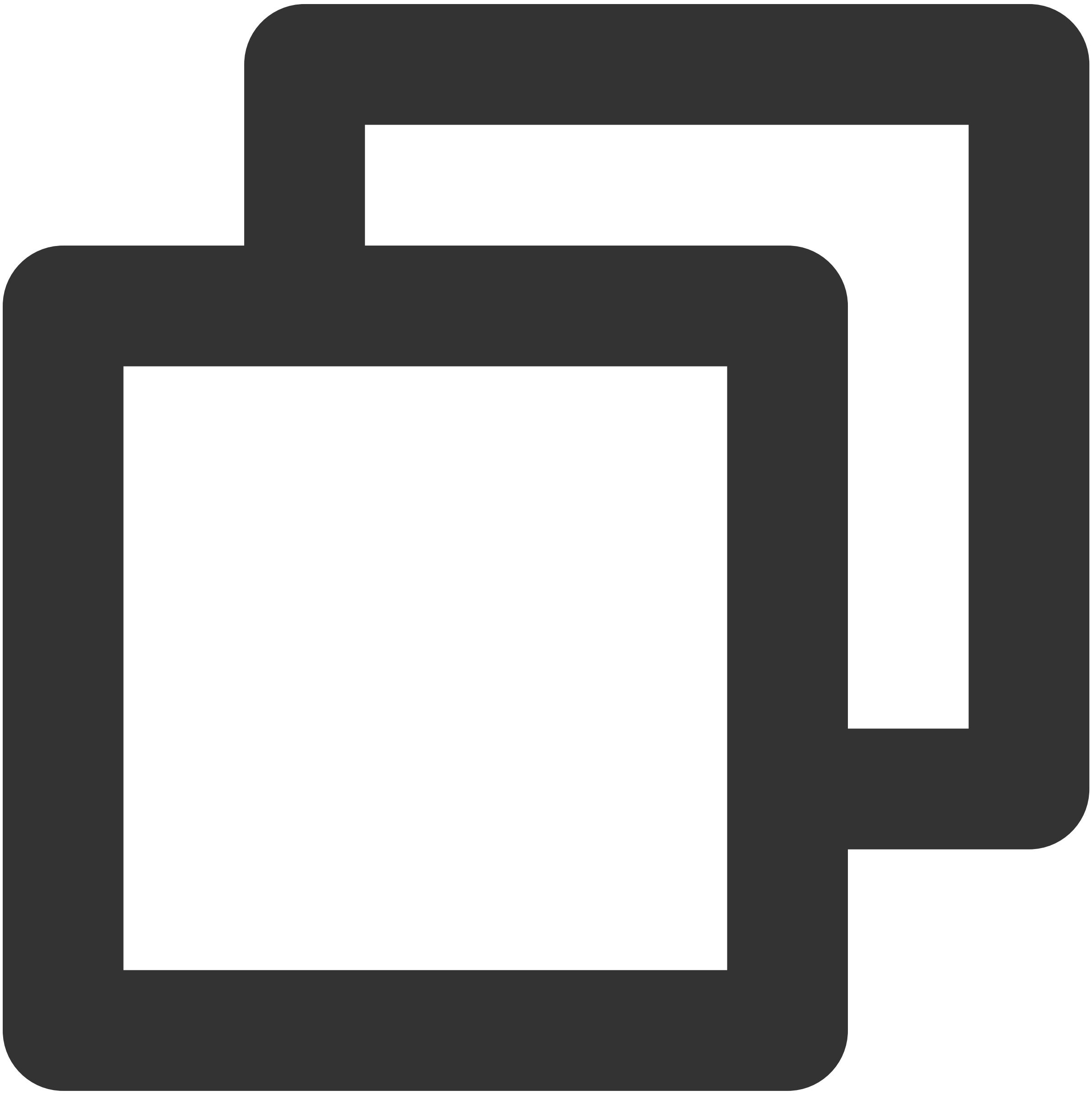
Switch microphone



```
trtc.updateLocalAudio({ option: { microphoneId }});
```

Device plug and unplug detection

Release microphone usage after detection is complete



```
trtc.stopLocalAudio();
```

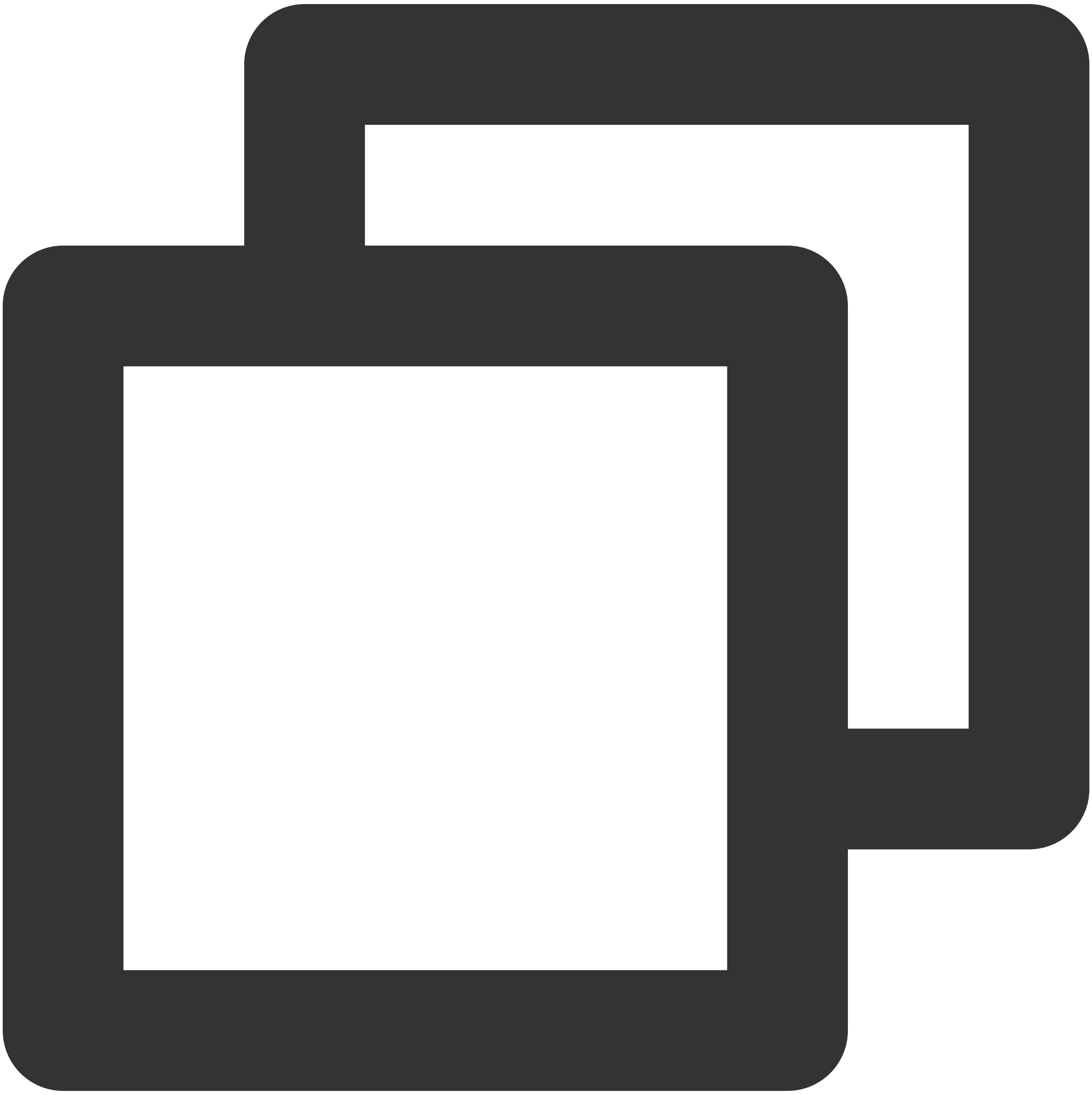
4) Speaker Detection

Detection principle: Play an mp3 media file through the audio tag.

Create an audio tag to remind the user to turn up the volume and play the mp3 to confirm whether the speaker device is working properly.


```
<audio id="audio-player" src="xxxxx" controls></audio>
```

Stop playing after detection is complete



```
const audioPlayer = document.getElementById('audio-player');
if (!audioPlayer.paused) {
    audioPlayer.pause();
}
audioPlayer.currentTime = 0;
```

5) Network Detection

Reference: Network Quality Detection Before Call

TRTC Capability Detection Page

You can use the TRTC Detection Page where you are currently using the TRTC SDK to detect the current environment. You can also click the "Generate Report" button to get a report of the current environment for environment detection or troubleshooting.

Testing Network Quality Android&iOS&Windows&Mac

Last updated：2024-06-07 22:18:34

It is difficult for ordinary users to measure network quality. Before calls are made, we recommend that you test the network speed to get more accurate results on network quality.

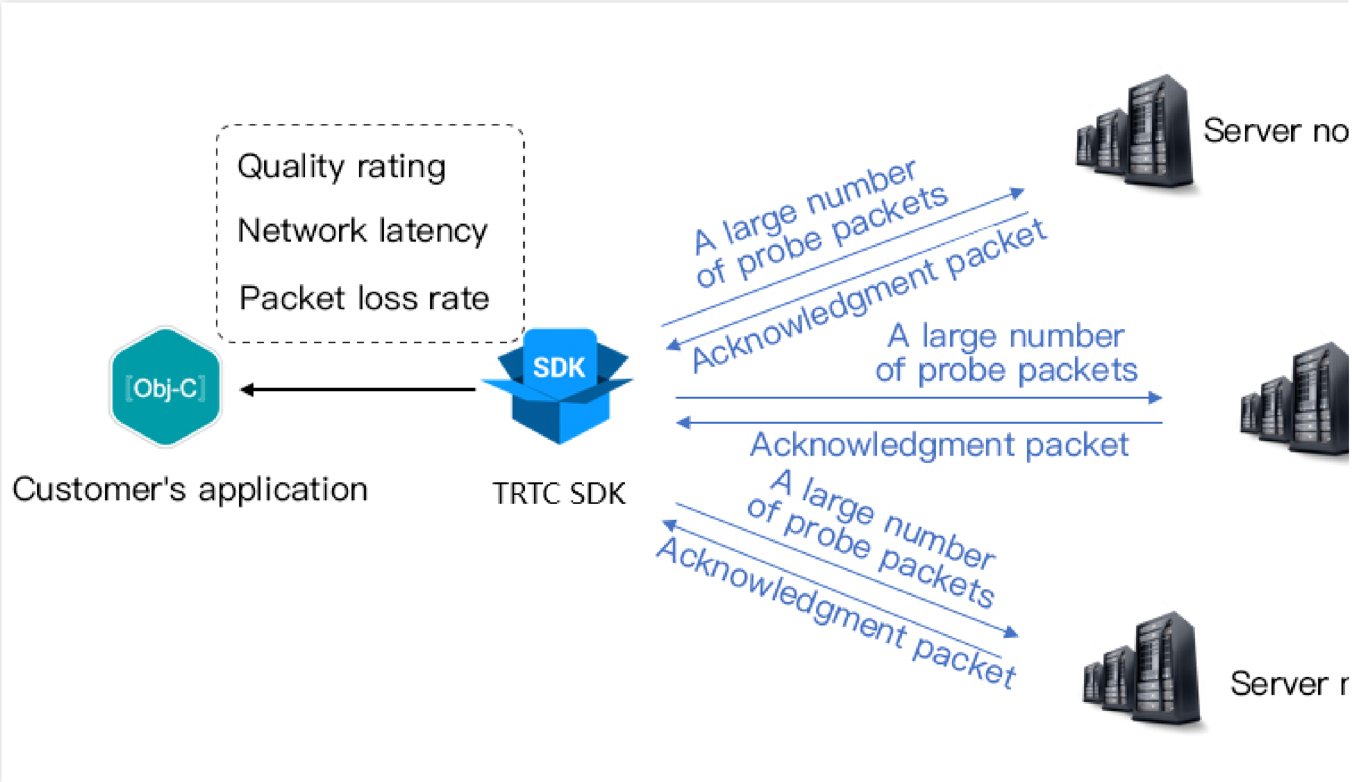
Notes

To ensure call quality, do not run the test during a video call.
Speed testing consumes traffic and consequently generates a small traffic fee (almost negligible).

Supported Platforms

iOS	Android	macOS	Windows	Electron	Web
✓	✓	✓	✓	✓	✓ (Reference: Detect Network Quality)

How Speed Testing Works



During speed testing, the SDK sends a batch of probe packets to the server node, measures the quality of return packets, and returns the testing result via a callback API.

The testing result can be used to optimize the SDK's server selection policy, so you are advised to run the test before the first call, which will help the SDK select the optimal server. If the result is unsatisfactory, you can show a UI message asking users to change to a better network.

The test result (TRTCSpeedTestResult) includes the following parameters:

Parameter	Type	Description
success	Success result	Whether the test is successful.
errMsg	Error message	Error message of bandwidth test.
ip	Server address	Testing server IP
quality	Network quality score	Network quality measured by the SDK. Lower packet loss and shorter RTT result in a higher network quality score.
upLostRate	Upstream packet loss rate	Value range: 0-1.0. `0.3` indicates that for every 10 data packets sent to the server, 3 may be lost.

downLostRate	Downstream packet loss rate	Value range: 0-1.0. `0.2` indicates that for every 10 data packets received from the server, 2 may be lost.
rtt	Latency	The time it takes for data to travel from the SDK to the server and back again. The shorter the RTT, the better. The normal range of RTT is 10-100 ms.
availableUpBandwidth	Upstream bandwidth	Estimated upstream bandwidth in Kbps. -1 indicates an invalid value.
availableDownBandwidth	Downstream bandwidth	Estimated downstream bandwidth in Kbps. -1 indicates an invalid value.

How to Test Speed

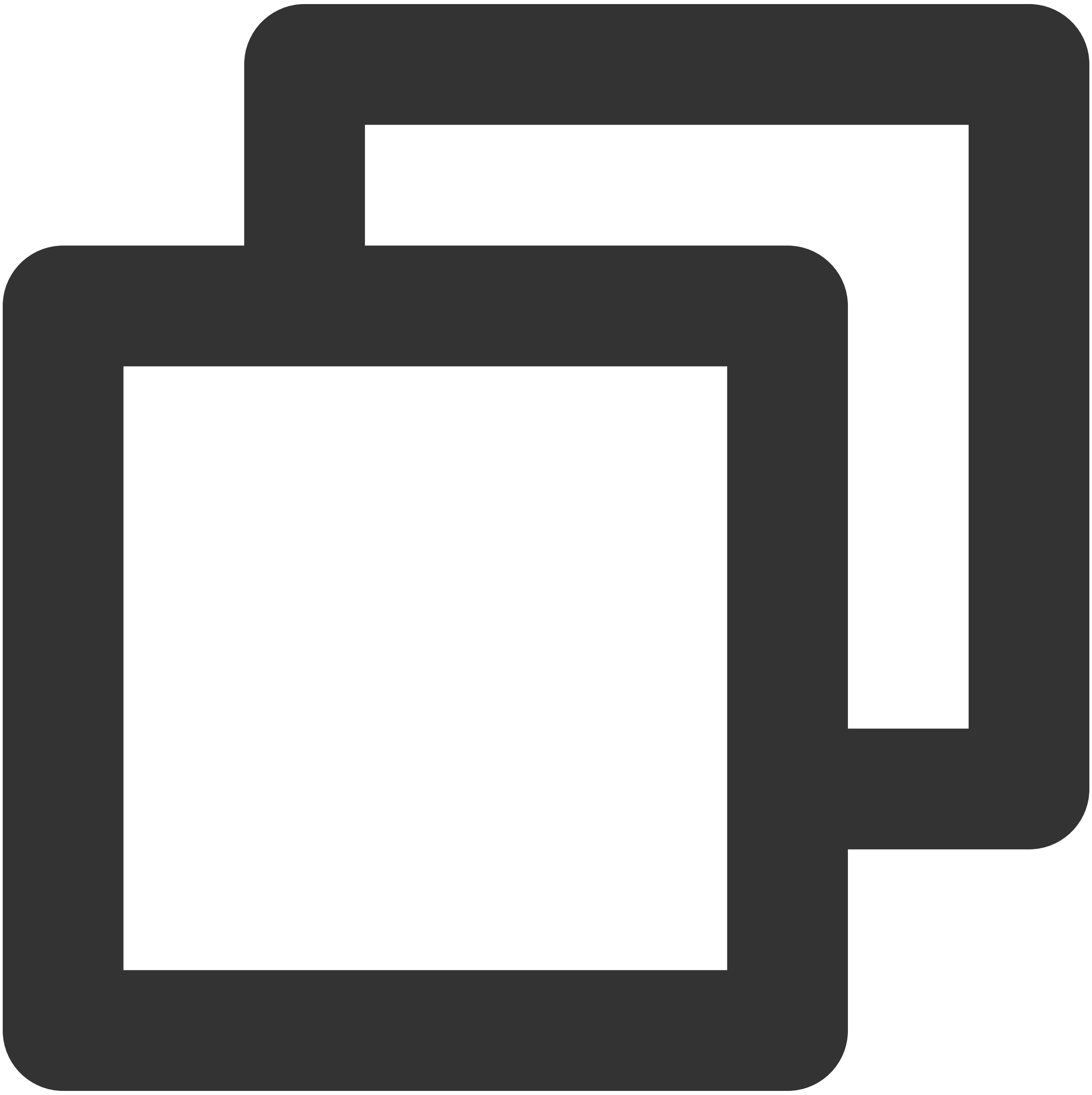
The speed test feature can be started through the `startSpeedTest` function of `TRTCCloud` . The speed test result will be called back through the callback function.

Objective-C

Java

C++

C#



```
// Sample code for starting speed testing. `sdkAppId` and `UserSig` are required. F
// The example below starts after login.
- (void)onLogin:(NSString *)userId userSig:(NSString *)userSid
{
    TRTCSpeedTestParams *params;
    // `sdkAppID` is the actual application ID obtained from the console.
    params.sdkAppId = sdkAppId;
    params.userID = userId;
    params.userSig = userSig;
    // Expected upstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not t
    params.expectedUpBandwidth = 5000;
```

```
    // Expected downstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not
    params.expectedDownBandwidth = 5000;
    [trtcCloud startSpeedTest:params];
}
- (void)onSpeedTestResult:(TRTCSpeedTestResult *)result {
    // The speed test result will be called back after the test is completed
}
```

```
// Sample code for starting speed testing. `sdkAppId` and `UserSig` are required. F
// The example below starts after login.
```



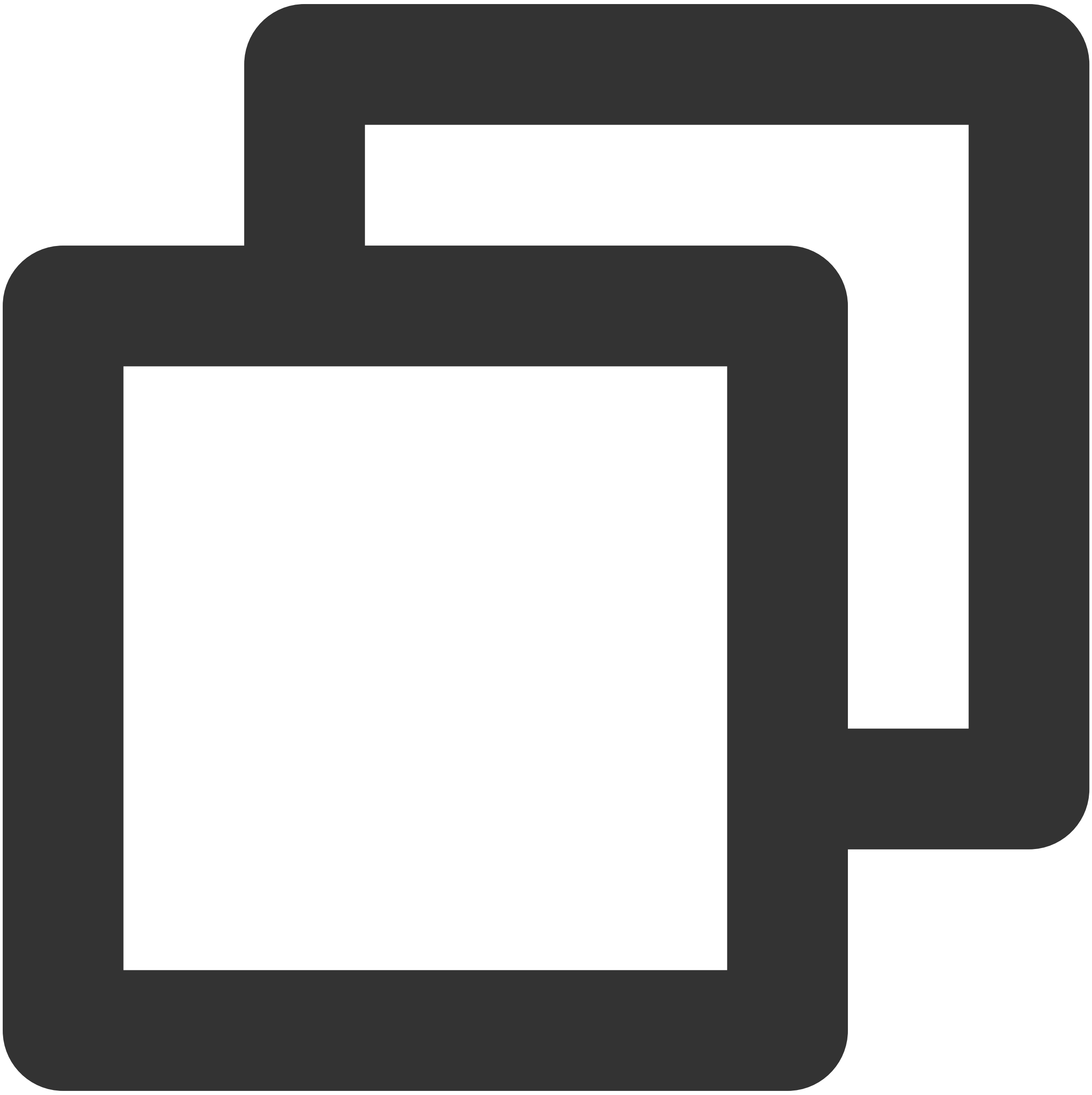
```
public void onLogin(String userId, String userSig)
{
  TRTCCloudDef.TRTCSpeedTestParams params = new TRTCCloudDef.TRTCSpeedTestParams();
  params.sdkAppId = GenerateTestUserSig.SDKAPPID;
  params.userId = mEtUserId.getText().toString();
  params.userSig = GenerateTestUserSig.genTestUserSig(params.userId);
  params.expectedUpBandwidth = Integer.parseInt(expectUpBandwidthStr);
  params.expectedDownBandwidth = Integer.parseInt(expectDownBandwidthStr);
    // `sdkAppID` is the actual application ID obtained from the console.
    trtcCloud.startSpeedTest(params);
}

// Listen for the test result. Inherit `TRTCCloudListener` and implement the follow
void onSpeedTestResult(TRTCCloudDef.TRTCSpeedTestResult result)
{
    // The speed test result will be called back after the test is completed
}
```

```
// Sample code for starting speed testing. `sdkAppId` and `UserSig` are required. F
// The example below starts after login.
void onLogin(const char* userId, const char* userSig)
{
    TRTCSpeedTestParams params;
    // `sdkAppID` is the actual application ID obtained from the console.
    params.sdkAppId = sdkAppId;
    params.userId = userid;
    param.userSig = userSig;
    // Expected upstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not t
    param.expectedUpBandwidth = 5000;
```

```
    // Expected downstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not
    param.expectedDownBandwidth = 5000;
    trtcCloud->startSpeedTest(params);
}

// Listen for the testing result
void TRTCCloudCallbackImpl::onSpeedTestResult(
             const TRTCSpeedTestResult& result)
{
    // The speed test result will be called back after the test is completed
}
```



```
// Sample code for starting speed testing. `sdkAppId` and `UserSig` are required. F
// The example below starts after login.
private void onLogin(string userId, string userSig)
{
    TRTCSpeedTestParams params;
    // `sdkAppID` is the actual application ID obtained from the console.
    params.sdkAppId = sdkAppId;
    params.userId = userid;
    param.userSig = userSig;
    // Expected upstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not t
    param.expectedUpBandwidth = 5000;
```

```
    // Expected downstream bandwidth in Kbps. Value range: 10-5000. 0 indicates not
    param.expectedDownBandwidth = 5000;
    mTRTCCloud.startSpeedTest(params);
}

// Listen for the testing result
public void onSpeedTestResult(TRTCSpeedTestResult result)
{
    // The speed test result will be called back after the test is completed
}
```

Speed Test Tool

If you don't want to call an API to test the network speed, you can use the network speed test tool for PC provided by TRTC to quickly get the network quality details.

Download link

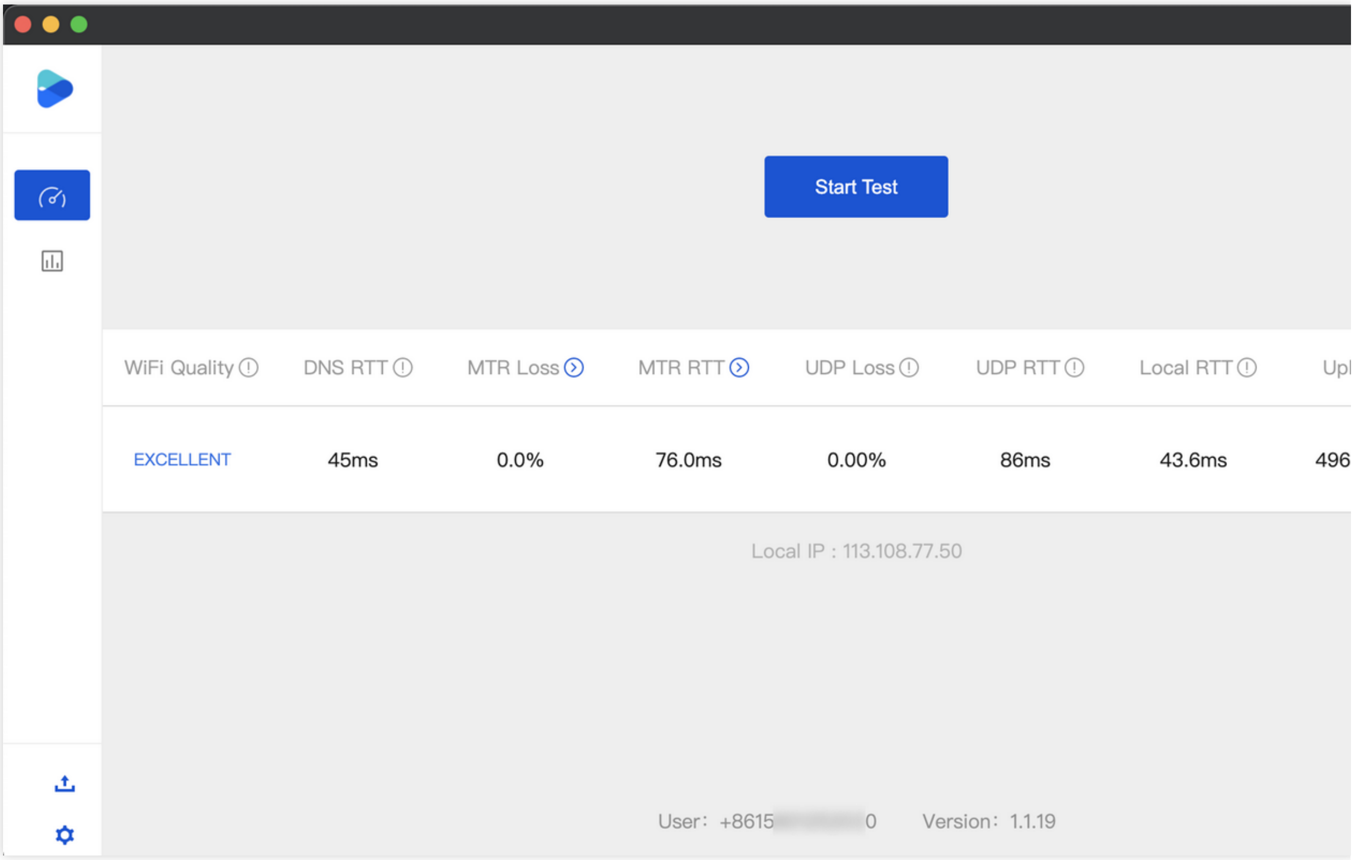
Mac | Windows

Test metrics

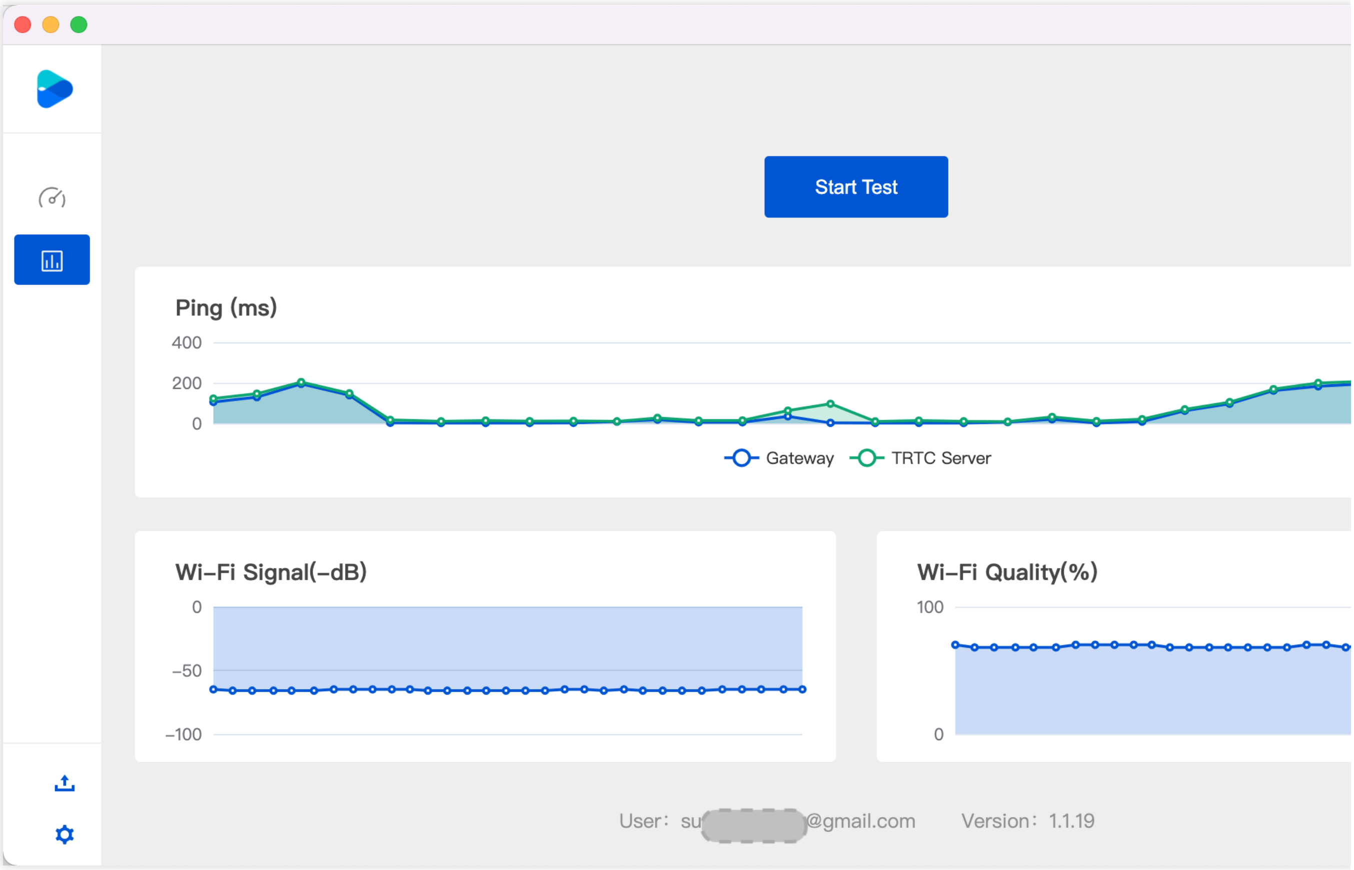
Metric	Description
WiFi Quality	Wi-Fi signal reception quality
DNS RTT	Tencent Cloud testing domain DNS round-trip time (RTT)
MTR	MTR is a network speed test tool, which can detect the packet loss rate and latency between client and TRTC node and display the details of each hop in the route
UDP Loss	UDP packet loss rate between client and TRTC node
UDP RTT	UDP latency between client and TRTC node
Local RTT	Latency between client and local gateway
Upload	Estimated upstream bandwidth
Download	Estimated downstream bandwidth

Tool screenshots

Quick test:



Continuous test:


Start Test
Ping (ms)
400
200
0
Gateway
TRTC Server
Wi-Fi Signal(-dB)
0
-50
-100
Wi-Fi Quality(%)
100
0
User：su
@gmail.com
Version：1.1.19

Web

Last updated：2023-09-28 11:54:53

Before entering the room or during the call, you can check the user's network quality to determine the current network quality. If the user's network quality is too poor, it is recommended that the user change the network environment to ensure normal call quality.

This article mainly introduces how to implement network quality detection before the call based on the `NETWORK_QUALITY` event.

Network quality check during the call process

```
const trtc = TRTC.create();
trtc.on(TRTC.EVENT.NETWORK_QUALITY, event => {
   console.log(`network-quality, uplinkNetworkQuality:${event.uplinkNetworkQuality}
   console.log(`uplink rtt:${event.uplinkRTT} loss:${event.uplinkLoss}`)
   console.log(`downlink rtt:${event.downlinkRTT} loss:${event.downlinkLoss}`)
})
```

Network quality check before making a call

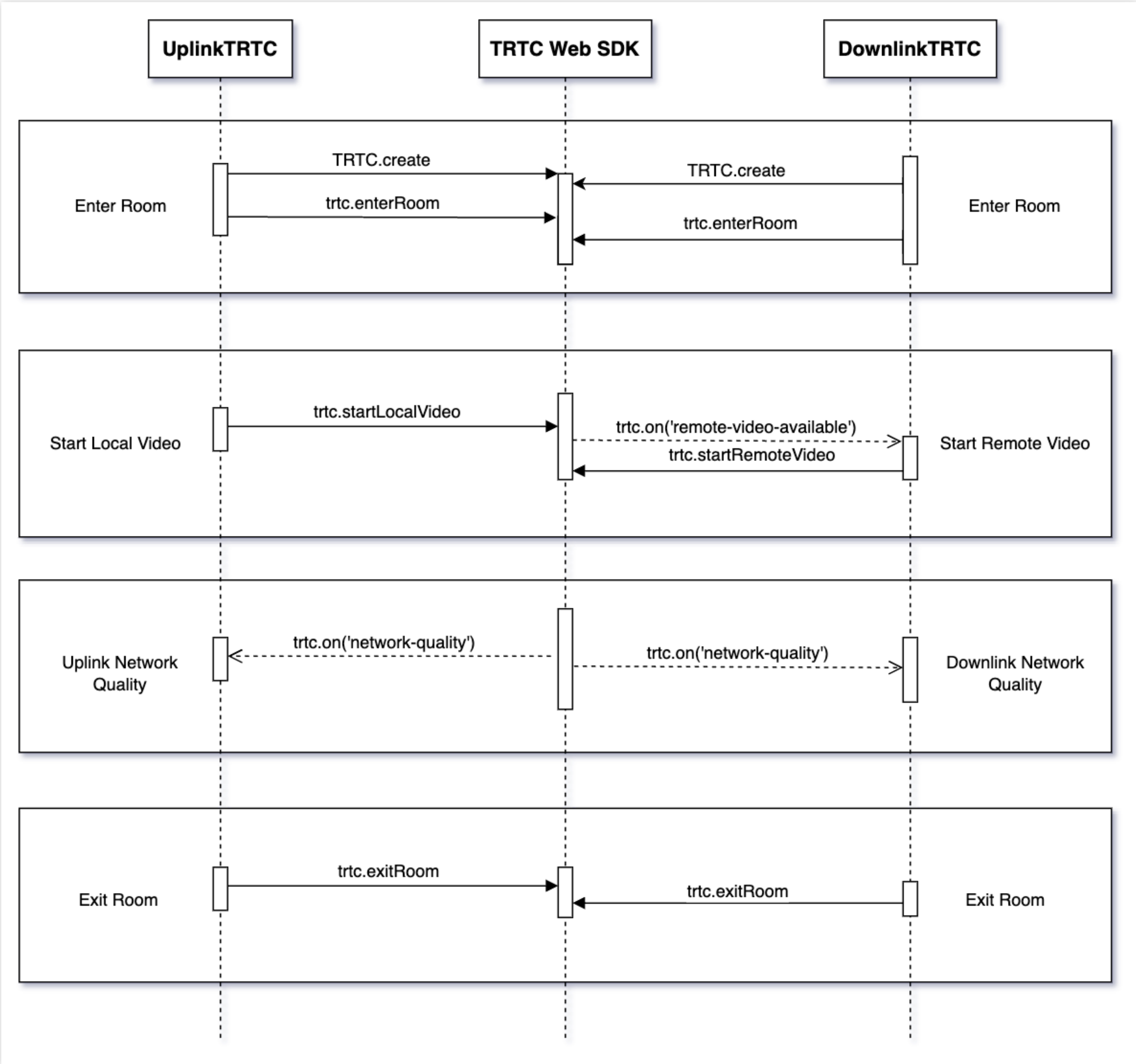
Implementation process

1. Call `TRTC.create()` to create two TRTCs, referred to as uplinkTRTC and downlinkTRTC.
2. Both TRTCs enter the same room.
3. Use uplinkTRTC to push the stream, and listen to the `NETWORK_QUALITY` event to detect the uplink network quality.
4. Use downlinkTRTC to pull the stream, and listen to the `NETWORK_QUALITY` event to detect the downlink network quality.
5. The entire process lasts for about 15 seconds, and finally takes the average network quality to roughly determine the uplink and downlink network conditions.

notice

The process of checking network quality incurs a small basic service fee. If a resolution is not specified, the stream will be published at a resolution of 640 x 480.

API Call Sequence


UplinkTRTC
TRTC Web SDK
DownlinkTRTC
Enter Room
TRTC.create
trtc.enterRoom
TRTC.create
trtc.enterRoom
Enter Room
Start Local Video
trtc.startLocalVideo
trtc.on('remote-video-available')
trtc.startRemoteVideo
Start Remote Video
Uplink Network Quality
trtc.on('network-quality')
trtc.on('network-quality')
Downlink Network Quality
Exit Room
trtc.exitRoom
trtc.exitRoom
Exit Room

Sample Code

```
let uplinkTRTC = null; // Used to detect uplink network quality
let downlinkTRTC = null; // Used to detect downlink network quality
let localStream = null; // Stream used for testing
let testResult = {
  // Record uplink network quality data
  uplinkNetworkQualities: [],
  // Record downlink network quality data
  downlinkNetworkQualities: [],
  average: {
    uplinkNetworkQuality: 0,
```

```
    downlinkNetworkQuality: 0
  }
}

// 1. Test uplink network quality
async function testUplinkNetworkQuality() {
  uplinkTRTC = TRTC.create();

  uplinkTRTC.enterRoom({
    roomId: 8080,
    sdkAppId: 0, // Fill in sdkAppId
    userId: 'user_uplink_test',
    userSig: '', // userSig of uplink_test
    scene: 'rtc'
  })

  uplinkTRTC.on(TRTC.EVENT.NETWORK_QUALITY, event => {
    const { uplinkNetworkQuality } = event;
    testResult.uplinkNetworkQualities.push(uplinkNetworkQuality);
  });

}

// 2. Detect downlink network quality
async function testDownlinkNetworkQuality() {
  downlinkTRTC = TRTC.create();
  downlinkTRTC.enterRoom({
    roomId: 8080,
    sdkAppId: 0, // Fill in sdkAppId
    userId: 'user_downlink_test',
    userSig: '', // userSig
    scene: 'rtc'
  });

  downlinkTRTC.on(TRTC.EVENT.NETWORK_QUALITY, event => {
      const { downlinkNetworkQuality } = event;
      testResult.downlinkNetworkQualities.push(downlinkNetworkQuality);
  })
}

// 3. Start detection
testUplinkNetworkQuality();
testDownlinkNetworkQuality();

// 4. Stop detection after 15s and calculate the average network quality
setTimeout(() => {
  // Calculate the average uplink network quality
```

```
    if (testResult.uplinkNetworkQualities.length > 0) {
      testResult.average.uplinkNetworkQuality = Math.ceil(
        testResult.uplinkNetworkQualities.reduce((value, current) => value + current,
      );
    }

    if (testResult.downlinkNetworkQualities.length > 0) {
      // Calculate the average downlink network quality
      testResult.average.downlinkNetworkQuality = Math.ceil(
        testResult.downlinkNetworkQualities.reduce((value, current) => value + curren
      );
    }

    // Detection is over, clean up related states.
    uplinkTRTC.exitRoom();
    downlinkTRTC.exitRoom();
  }, 15 * 1000);
```

Result Analysis

After the above steps, you can get the average uplink network quality and the average downlink network quality. The enumeration values of network quality are as follows:

Value	Meaning
0	The network condition is unknown, indicating that the current TRTC instance has not established an uplink/downlink connection
1	The network condition is excellent
2	The network condition is good
3	The network condition is average
4	The network condition is poor
5	The network condition is extremely poor
6	The network connection has been disconnected. Note: If the downlink network quality is this value, it means that all downlink connections have been disconnected.

Suggestion:

When the network quality is greater than 3, it is recommended to guide the user to check the network and try to change the network environment, otherwise it is difficult to ensure normal audio and video communication.
 You

can also reduce bandwidth consumption through the following strategies:
If the uplink network quality is greater than 3, you can reduce the bitrate through the `TRTC.updateLocalVideo()` interface or close the video through the `TRTC.stopLocalVideo()` method to reduce uplink bandwidth consumption.
If the downlink network quality is greater than 3, you can reduce the downlink bandwidth consumption by subscribing to a small stream (refer to: Enable Small Stream Transmission) or only subscribing to audio.

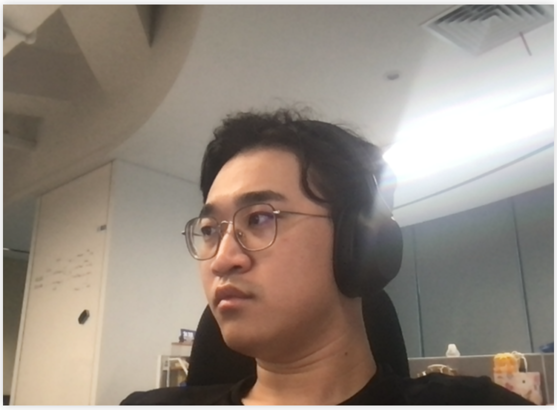
Utilizing Virtual Backgrounds

Web

Last updated：2024-07-03 16:15:44

Function Description

This document will describe how to implement the virtual background feature during a call.

Original Camera	Background Blur
	

Prerequisites

The SdkAppId that requires the use of the virtual background feature has subscribed to the RTC-Engine Monthly Packages, Pro edition (1499 USD/month).

TRTC SDK version must be ≥ 5.5.0

System and configuration requirements for various platforms are as follows:

Platform	Operating System	Browser Version	FPS	Recommended Configuration	Remarks
Web	Windows	Chrome 90+Firefox 90+Edge 97+	30	Memory: 16GBCPU: i5-10500GPU: Dedicated 2GB	It is recommended to use the latest version of
			15	Memory: 8GBCPU: i3-	

				8300GPU: Integrated Intel 1GB	Chrome browser (Enable hardware acceleration in the browser)
	Mac	Chrome 98+Firefox 96+Safari 14+	30	2019 MacBookMemory: 16GB (2667MHz)CPU: i7 (6-core 2.60GHz)GPU: AMD Radeon 5300M	
	Android	ChromeFirefox Browser	30	High-end devices (e.g., Qualcomm Snapdragon 8 Gen1)	It is recommended to use Chrome, or Firefox browsers and other mainstream browsers
			20	Mid-range devices (e.g., MediaTek Dimensity 8000-MAX)	
			10	Low-end devices (e.g., Qualcomm Snapdragon 660)	
	iOS	ChromeSafariFirefox	30	iPhone 13	- Requires iOS 14.4 or above- It is recommended to use Chrome or Safari browsers
			20	iPhone XR	

Implementation Steps

Registering the Plugin

```
import { VirtualBackground } from 'trtc-sdk-v5/plugins/video-effect/virtual-backgro

let trtc = TRTC.create({ plugins: [VirtualBackground] });
```

Starting the Local Camera

```
await trtc.startLocalVideo();
```

Enabling the Virtual Background Plugin

```
await trtc.startPlugin('VirtualBackground', {
  sdkAppId: 123123,
  userId: 'userId_123',
  userSig: 'your_userSig'
});
```

Note:

The first time you enable this feature, it requires loading computing resources, which has been automatically processed for you. For a more ultimate startup experience, please refer to the end of the document: FAQ > How to speed up startup?

Updating Parameters as Needed

```
// Changing to an image background
await trtc.updatePlugin('VirtualBackground', {
  type: 'image',
  src: 'https://picsum.photos/seed/picsum/200/300'
});
```

Disabling Virtual Background


```
await trtc.stopPlugin('VirtualBackground');
```

API Documentation

trtc.startPlugin('VirtualBackground', options)

Used to enable the virtual background.

options

Name	Type	Attributes	Description
sdkAppId	`number`	Required	Current application ID
userId	`string`	Required	Current user ID
userSig	`string`	Required	UserSig corresponding to the user ID
type	`string`	Optional	`image` for image background `blur` for background blur (default)
src	`string`	Required if type is `image`	Image URL, such as `https://picsum.photos/seed/picsum/200/300`
onError	`(error) => {}`	Optional	Callback for errors that occur during runtime error.code=10000003 indicates high rendering latency error.code=10000006 indicates insufficient browser feature support, which may lead to lagging. Recommended solutions can be found in the Common Issues section at the end of the document.

Example:

```
await trtc.startPlugin('VirtualBackground', {
  sdkAppId: 123123,
  userId: 'userId_123',
  userSig: 'your_userSig',
  type: 'image',
  src: 'https://picsum.photos/seed/picsum/200/300'
});
```

trtc.updatePlugin('VirtualBackground', options)

Allows modification of virtual background parameters.

options

Name	Type	Attributes	Description
type	`string`	Required	`image` for image background `blur` for background blur
src	`string`	Required if type is `image`	Image URL, such as `https://picsum.photos/seed/picsum/200/300`

Example:

```
await trtc.updatePlugin('VirtualBackground', {
  type: 'blur'
});
```

trtc.stopPlugin('VirtualBackground')

Disables the virtual background.

Frequently Asked Questions

When running the demo in Chrome, the video appears upside down and laggy?

This plugin uses GPU acceleration, and you need to enable hardware acceleration mode in your browser settings. You can copy and paste `chrome://settings/system` into your browser's address bar and enable hardware acceleration mode.

When device performance is insufficient and causes high latency with long rendering times?

You can reduce video resolution or frame rate by listening to events.

```
async function onError(error) {
  const { code } = error;
  if (code === 10000003 || code === 10000006) {
    // Reduce resolution and frame rate
    await trtc.updateLocalVideo({
      option: {
        profile: '480p_2'
      },
    });
    // await trtc.stopPlugin('VirtualBackground'); // Or disable the plugin
  }
}

await trtc.startPlugin('VirtualBackground', {
  ...// Other parameters
  onError,
});
```

How to start the plugin faster? (Manually deploy virtual background model files)

The principle of this plugin is: after capturing from the camera, perform portrait segmentation locally through **machine learning models**, and then upload the results to the backend.

When first enabled, it will automatically download machine learning models from Tencent Cloud CDN, but Tencent Cloud CDN and the static file address of your webpage are not from the same-origin, which will reduce some loading speeds. **The best solution is to deploy model files on your own static resource server, such as deploying them together with web services.**

The acceleration method is very simple, unzip assets.zip into your project, such as `./src/assets/` , and specify this address when creating a TRTC instance:

```
const trtc = TRTC.create({ plugins: [VirtualBackground], assetsPath: './src/assets/
```

On-Cloud Recording

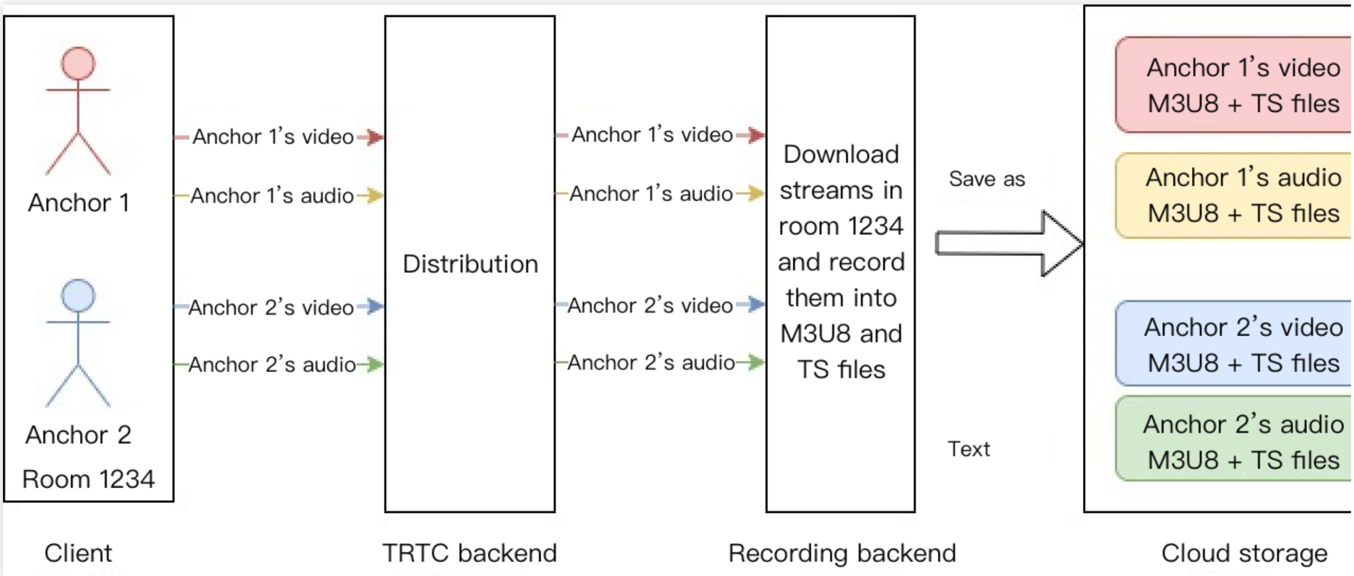
Last updated：2024-04-26 17:43:17

For scenarios such as online education, live showroom, video conferencing, online medical consultation, and remote banking, it is often necessary to record entire video calls or live streaming sessions for purposes including content moderation, archiving, and playback. The on-cloud recording feature of TRTC can help meet these demands.

Overview

The on-cloud recording feature of TRTC allows you to record an audio/video stream in real time using a RESTful API. It is flexible, light, and easy-to-use, saving you the trouble of deploying servers and recording modules.
Recording mode: Single-stream recording records the audio and video of each user in a room separately, while mixed-stream recording records all audios and videos in a room into one result.
Stream subscription: You can determine whose streams you receive or do not receive using an allowlist/blocklist.
Transcoding parameters: In the mixed-stream recording mode, you can determine the output video quality by specifying transcoding parameters.
Stream-mixing parameters: For mixed-stream recording, we offer multiple auto-arranged layout templates. You can also customize a layout template.
File storage: Currently, you can save recording files only in Tencent Cloud COS or VOD.
(We plan to add support for storage and video-on-demand services of third-party cloud vendors in the future. To save files to third-party platforms, you will need to provide your cloud service account and the storage parameters.)
Callback notification: By configuring a callback domain in the console, you can receive notifications about on-cloud recording events via your callback server.

Single-stream recording

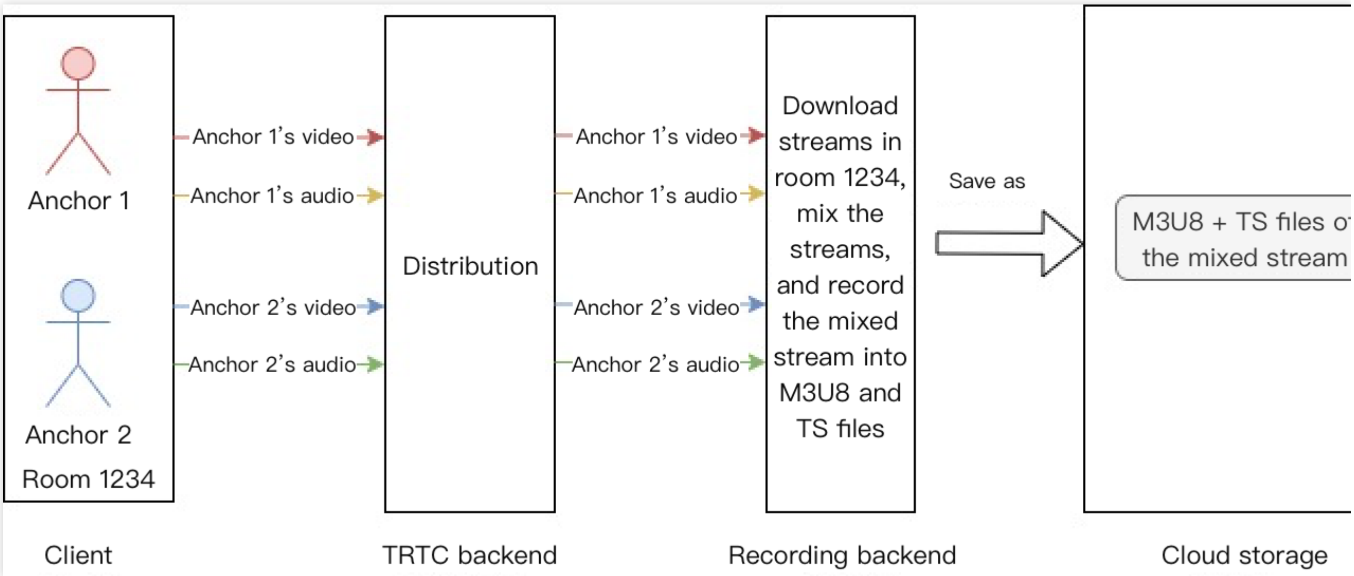


The diagram above shows the workflow of single-stream recording. In room 1234, anchor 1 and anchor 2 are publishing streams. If you subscribe to their streams and enable single-stream recording, the TRTC backend will record the audio and video data of anchor 1 and anchor 2 separately. The recording results will include:

1. An M3U8 index file of anchor 1's video
2. Multiple TS segment files of anchor 1's video
3. An M3U8 index file of anchor 1's audio
4. Multiple TS segment files of anchor 1's audio
5. An M3U8 index file of anchor 2's video
6. Multiple TS segment files of anchor 2's video
7. An M3U8 index file of anchor 2's audio
8. Multiple TS segment files of anchor 2's audio

The backend will then upload the files to the cloud storage server you specify. You can download the files and merge/transcode them. We offer a script for merging audio and video streams.

Mixed-stream recording



The above shows the workflow of mixed-stream recording. In room 1234, anchor 1 and anchor 2 are publishing streams. If you subscribe to their streams and enable mixed-stream recording, the TRTC backend will mix the streams of anchor 1 and anchor 2 according to the layout template you specify and then record them into one result, which will include:

1. An M3U8 index file of the mixed video
2. Multiple TS segment files of the mixed video

The backend will then upload the files to the cloud storage server you specify. You can download the files and merge/transcode them. We offer a script for merging audio and video streams.

Note:

The rate limit for the recording API is 20 calls per second.

The timeout period for a query is six seconds.

We allow up to 100 ongoing recording tasks at the same time. If you need to record more, please submit a ticket.

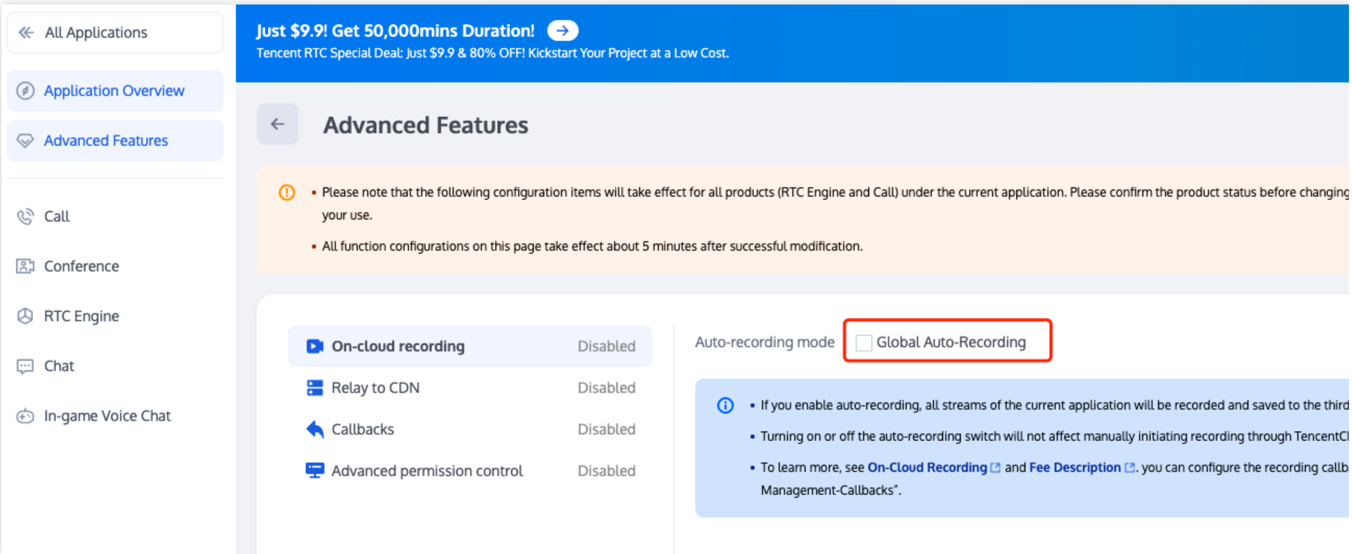
In the single-stream recording mode, you can record up to 25 streams in a room at the same time.

Auto-Recording

TencentRTC offers an auto-recording feature, eliminating the need for manual recording task management. To use this recording solution, go to **Console** > Applications, select the desired application, and click

on the right side to enter the application information page. Then click **Advanced Features**, enable on-cloud auto-recording, complete the global auto-recording template configuration, and submit it.

After it becomes effective (wait for 5-10 minutes for it to take effect), the publishing of audio and video by the anchors in the TRTC room will trigger the start of the recording task. The recording task will be triggered to stop after all the anchors have left the room and the set timeout period for resumption has been exceeded.

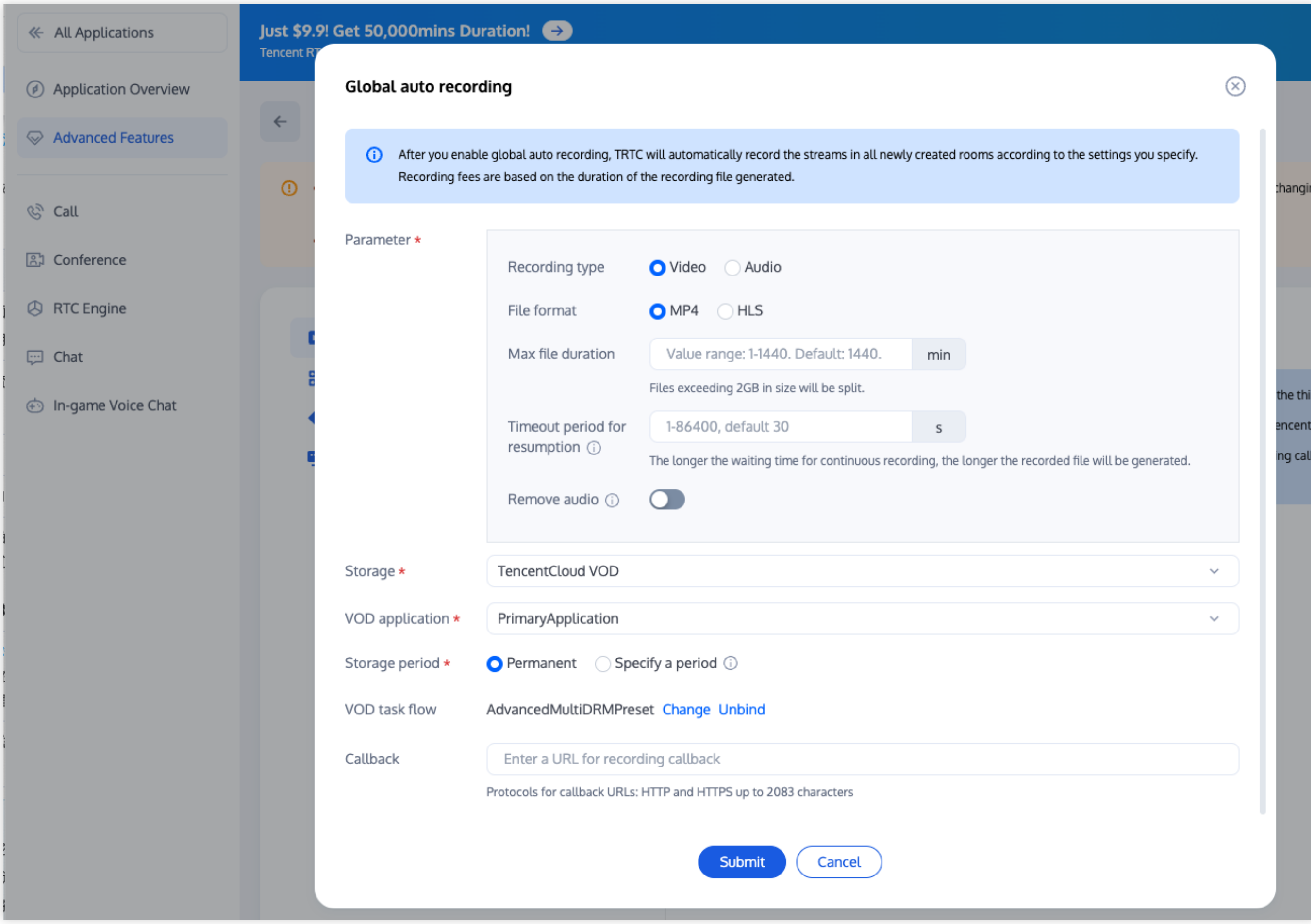


Before enabling the global auto-recording feature, configure the global auto-recording template. Global auto-recording supports single-stream recording (i.e., record a separate file for each anchor), and once enabled, it is only effective for newly created rooms. It does not apply to rooms created before the auto-recording feature was enabled.

Global single-stream recording supports audio and video recording, audio-only recording, and video-only recording. The recording file formats supported are MP4, HLS, and AAC (in audio-only recording format).

Configuration Item	Description
Recording mode	Single-stream recording: The video data of each anchor in the room will be saved as a separate file. To record the mixed video of multiple anchors, use Manual Merge Recording.
Recording type	Audio and video: Both audio and video streams in the room are recorded, suitable for video calls and interactive live streaming scenarios. Audio-only: Only audio streams in the room are recorded.
File format	Supports MP4, HLS, and AAC (in audio-only format).
Max file duration	Specifies the segment duration of the recording file, with a range of 1-1,440 minutes. The default is 1,440 minutes.
Timeout period for resumption	Sets the timeout period for resuming recording in seconds. When the interruption interval does not exceed the set timeout period, a single call (or live stream) will generate only one file. However, the recording file will be received only after the timeout period for resumption expires. **The value range is 1 - 86,400 (default 30s)** . **Note: During the resumption waiting period, single-stream recording fees will be charged based on the audio duration. Please set it appropriately.**

Storage	Supports storage to Tencent Cloud Video on Demand (VOD), Tencent Cloud Object Storage (COS), and AWS S3 Storage. VOD: Requires support for specifying the VOD application and the storage period of recording files in VOD, and binding VOD task flows. COS & AWS: Requires completion of configuration for the corresponding bucket. Ensure you have the write permission to the bucket.
Callback address and callback key	The latest on-cloud recording service offers detailed recording event features. You can configure a server-side URL to receive recording callback events, and can also configure a callback key to verify the security of callback events. More information available here.



Note:

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

If the timeout period for resumption has not expired, the recording robot will continue to wait in the room for the anchor's publishing to complete the recording. The recording will not end immediately after the anchor leaves the room, so set the timeout period appropriately.

Single-stream recording can record the audio and video of up to 25 anchors in one room. If there are more than 25 anchors in the room, they will be sorted by the time they entered the room, and only the audio and video of the first 25 anchors will be recorded. (For single-stream recording of more than 25 anchors, see API recording).

Manual Recording Process

1. Start recording

Call the RESTful API `CreateCloudRecording` from your server to start on-cloud recording. Pay attention to the following parameters:

TaskId

This parameter uniquely identifies a recording task. Note it as you will need to provide it for other actions on the same task later.

RecordMode

Single-stream recording separately records the audios and videos of the anchors you subscribe to and uploads the recording files (including M3U8 and TS segment files) to the cloud.
Mixed-stream recording records all the audios and videos of the anchors you subscribe to into one result and uploads the recording files (including M3U8 and TS segment files) to the cloud.

SubscribeStreamUserIds

By default, on-cloud recording records all the streams (max 25) you receive in a room. You can use this parameter to specify whose streams you want to record and can change its value during recording.

StorageParams

You can use this parameter to specify the cloud storage/video-on-demand service you want to save recording files to. Make sure you use a valid value and that the cloud storage/video-on-demand service you use is available. Below are the naming conventions of recording files:

Naming of recording files

M3U8 file in the single-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>__UserId_s_<UserId>__UserId_e_<MediaId>_<Type>.m3u8
TS segment file in the single-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>__UserId_s_<UserId>__UserId_e_<MediaId>_<Type>_<UTC>.ts
MP4 file in the single-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>__UserId_s_<UserId>__UserId_e_<MediaId>_<Index>.mp4
M3U8 file in the mixed-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>.m3u8

TS segment file in the mixed-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>_<UTC>.ts
MP4 file in the mixed-stream recording mode:
<Prefix>/<TaskId>/<SdkAppId>_<RoomId>_<Index>.mp4
Naming of recovered files
The on-cloud recording feature has a high availability scheme that can recover recording files if the server fails. To prevent the recovered files from replacing the original files, we add a prefix `ha<1/2/3>` to the names of recovered files. The numbers indicate the times (max 3) the high availability scheme is used.
M3U8 file in the single-stream recording mode:
<Prefix>/<TaskId>/ha<1/2/3>_<SdkAppId>_<RoomId>__UserId_s_<UserId>__UserId_e_<MediaId>_<Type>.m3u8
TS segment file in the single-stream recording mode:
<Prefix>/<TaskId>/ha<1/2/3>_<SdkAppId>_<RoomId>__UserId_s_<UserId>__UserId_e_<MediaId>_<Type>_<UTC>.ts
M3U8 file in the mixed-stream recording mode:
<Prefix>/<TaskId>/ha<1/2/3>_<SdkAppId>_<RoomId>.m3u8
TS segment file in the mixed-stream recording mode:
<Prefix>/<TaskId>/ha<1/2/3>_<SdkAppId>_<RoomId>_<UTC>.ts

Field description

<Prefix>: The filename prefix. If this is not specified, the filename will not have a prefix.
<TaskId>: The task ID, which is unique and is returned by the start recording API.
<SdkAppId>: The application ID.
<RoomId>: The room ID.
<UserId>: The Base64-encoded ID of a user whose stream is recorded.
<MediaId>: Whether the primary stream (`main`) or substream (`aux`) is recorded.
<Type>: The type of stream that is recorded (`audio` or `video`).
<UTC>: The recording start time (UTC+0), which consists of the year, month, day, hours, minutes, seconds, and milliseconds.
<Index>: The index of a segment, which starts from 1. This field is used only if an MP4 file exceeds 2 GB or 24 hours and needs to be segmented.
ha<1/2/3>: The prefix for a file recovered by the high availability scheme. For example, if the scheme is used for the first time, the recovered file is named `<Prefix>/<TaskId>/ha1_<SdkAppId>_<RoomId>.m3u8` .

Note:

If \\<RoomId> is a string, it will be encoded into Base64. In the result, "/" is replaced with "-" and "=" is replaced with "."
<UserId> is encoded into Base64. In the result, "/" is replaced with "-" and "=" is replaced with "."

Recording start time

Recording starts when you start receiving data from an anchor. Recording start time is the Unix time on the server when recording starts.

You can query the start time of a recording task in three ways:

Via the `DescribeCloudRecording` API. The response parameter `BeginTimeStamp` indicates the recording start time (ms).

Below is a response of the `DescribeCloudRecording` API, in which `BeginTimeStamp` is `1622186279144`.

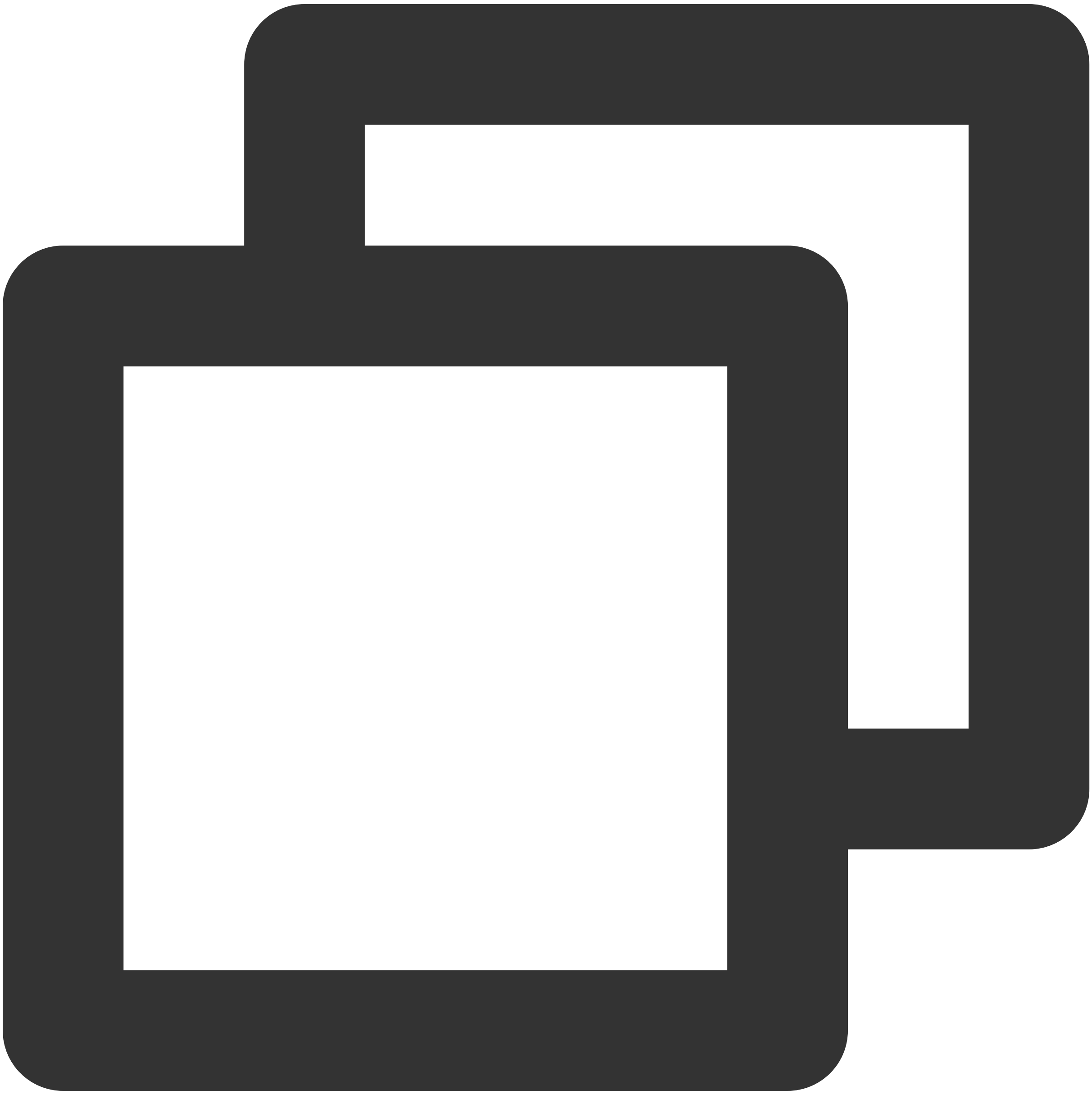
```
{
  "Response": {
```



```
      "Status": "xx",
      "StorageFileList": [
        {
          "TrackType": "xx",
          "BeginTimeStamp": 1622186279144,
          "UserId": "xx",
          "FileName": "xx"
        }
      ],
      "RequestId": "xx",
      "TaskId": "xx"
    }
  }
```

Via the #EXT-X-TRTC-START-REC-TIME directive of the M3U8 file

According to the M3U8 file below, the Unix time (ms) when recording started was 1622425551884.



```
#EXTM3U
#EXT-X-VERSION:3
#EXT-X-ALLOW-CACHE:NO
#EXT-X-MEDIA-SEQUENCE:0
#EXT-X-TARGETDURATION:70
#EXT-X-TRTC-START-REC-TIME:1622425551884
#EXT-X-TRTC-VIDEO-METADATA:WIDTH:1920 HEIGHT:1080
#EXTINF:12.074
1400123456_12345__UserId_s_MTY4NjExOQ..__UserId_e_main_video_20330531094551825.ts
#EXTINF:11.901
1400123456_12345__UserId_s_MTY4NjExOQ..__UserId_e_main_video_20330531094603825.ts
```

```
#EXTINF:12.076
1400123456_12345__UserId_s_MTY4NjExOQ..__UserId_e_main_video_20330531094615764.ts
#EXT-X-ENDLIST
```

Via a recording callback

If you have registered recording callbacks, you will receive a callback for the generation of recording files (event type 307), in which `BeginTimeStamp` indicates the recording start time.

According to the callback below, the Unix time (ms) when recording started was 1622186279144.

```
{
        "EventGroupId": 3,
```

```
        "EventType":    307,
        "CallbackTs":   1622186289148,
        "EventInfo":    {
                "RoomId":       "xx",
                "EventTs":      "1622186289",
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "FileName":     "xx.m3u8",
                        "UserId":       "xx",
                        "TrackType":    "audio",
                        "BeginTimeStamp":       1622186279144
                }
        }
}
```

MixWatermark

TRTC allows you to watermark videos during mixed-stream recording. You can add up to 25 watermarks to a video in your desired positions.

Field	Description
Top	The vertical offset of the watermark to the top left corner of the video.
Left	The horizontal offset of the watermark to the top left corner of the video.
Width	The watermark width.
Height	The watermark height.
url	The URL of the watermark file.

MixLayoutMode

TRTC supports the grid layout (default), floating layout, screen sharing layout, and custom layout.

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.

When there is only one video:

The video is scaled to fill the canvas.

When there are two videos:

The width of each video is half of the canvas width.

The height of each video is the same as the canvas height.

When there are three or four videos:

The canvas is split evenly into four windows and each video is displayed in one window.

When there are 5-9 videos:

The canvas is split evenly into nine windows and each video is displayed in one window.

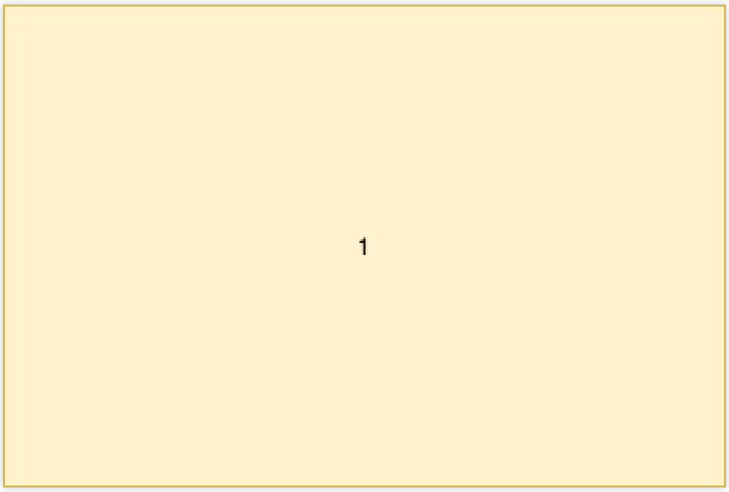
When there are 10-16 videos:

The canvas is split evenly into 16 windows and each video is displayed in one window.

When there are more than 16 videos:

The canvas is split evenly into 25 windows and each video is displayed in one window.

As shown below:



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.

When there are 17 or fewer videos:

The width and height of each small video are 23.5% of the canvas width and height.

The horizontal space and vertical space between two neighboring videos are 1.2% of the canvas width and height.

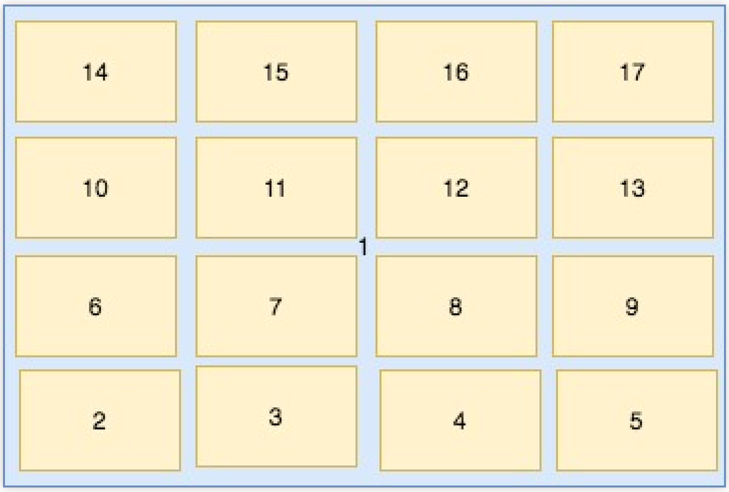
The right and left margins are 1.2% of the canvas width and the top and bottom margins are 1.2% of the canvas height.

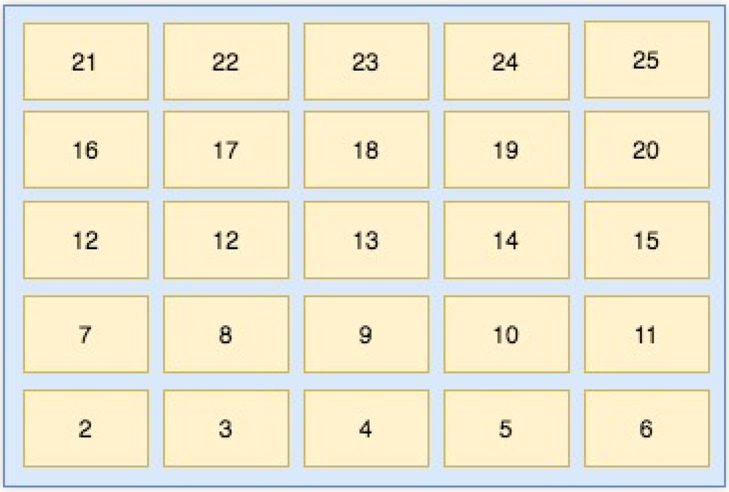
When there are more than 17 videos:

The width and height of each small video are 18.8% of the canvas width and height.

The horizontal space and vertical space between two neighboring videos are 1% of the canvas width and height.

The right and left margins are 1% of the canvas width and the top and bottom margins are 1% of the canvas height. As shown below:





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.

When there are five or fewer videos:

The size of each small video on the right is 1/5 the canvas width and 1/4 the canvas height.

The width of the large video on the left is 4/5 the canvas width, and its height is the same as the canvas height.

When there are six or seven videos:
The size of each small video on the right are 1/7 the canvas width and 1/6 the canvas height.
The width of the large video on the left is 6/7 the canvas width, and its height is the same as the canvas height.
When there are eight or nine videos:
The size of each small video on the right is 1/9 the canvas width and 1/8 the canvas height.
The width of the large video on the left is 8/9 the canvas width, and its height is the same as the canvas height.
When there are 10-17 videos:
The size of each small video on the right side is 1/10 the canvas width and 1/8 the canvas height.
The width of the large video on the left side is 4/5 the canvas width, and its height is the same as the canvas height.
When there are more than 17 videos:
The size of each small video on the right and bottom is 1/10 the canvas width and 1/8 the canvas height.
The width of the large video on the left is 4/5 the canvas width, and its height is 7/8 the canvas height.
As shown below:

Custom layout

You can also use `MixLayoutList` to customize a layout for anchor videos.

2. Query the recording task

You can use the `DescribeCloudRecording` API to query the status of an ongoing recording task. If the queried task has already ended, an error will be returned.
The file list (`StorageFile`) that is returned will include all the M3U8 files of the recording as well as the Unix time when recording started. If the task queried is a recording to VOD task, the `StorageFile` returned will be empty.

3. Modify recording parameters

You can use the `ModifyCloudRecording` API to modify recording parameters, including `SubscribeStreamUserIds` and `MixLayoutParams` (valid only for mixed-stream recording). Note that you need to specify all the parameters, including `MixLayoutParams` and `SubscribeStreamUserIds` , and not just the ones you want to modify. We recommend you note all the parameter values before a modification, or you will need to calculate them again.

4. Stop recording

You can call the `DeleteCloudRecording` API to stop recording. A recording task will also end automatically if there are no anchors (whether they are publishing data or not) in a room for longer than the specified time period (`MaxIdleTime`). We recommend you call the API to stop recording when you no longer need the service.

Advanced Features

Recording in MP4 format

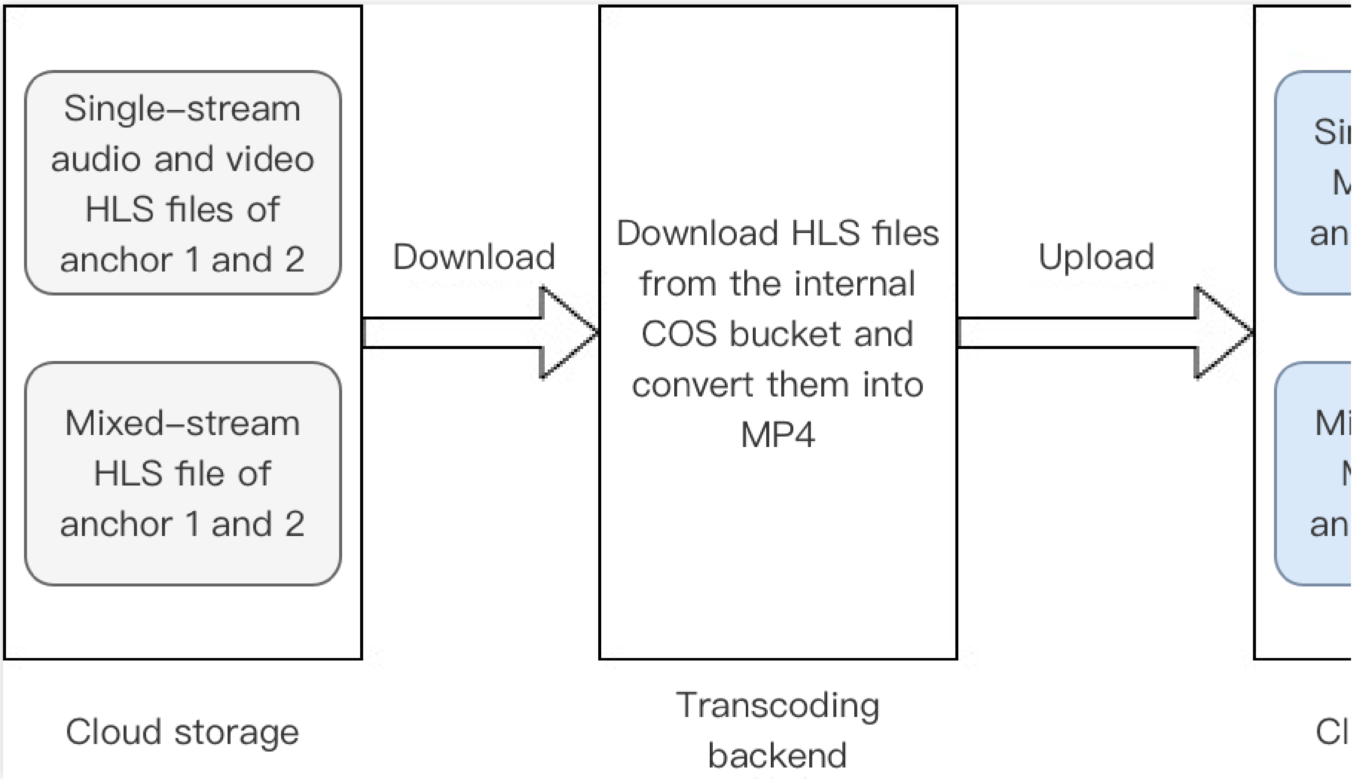
To record in MP4 format, set `OutputFormat` to `hls+mp4` when calling the `CreateCloudRecording` API. TRTC will still record in HLS format, but once recording ends, it will download the HLS file from the COS bucket where it is saved, convert it into MP4 format, and upload the MP4 file to the COS bucket.

Please note that COS download access is required for the above to work.

An MP4 file will be segmented if one of the following conditions is met.

1. The recording duration is longer than 24 hours.
2. The MP4 file exceeds 2 GB.

The workflow is as follows:

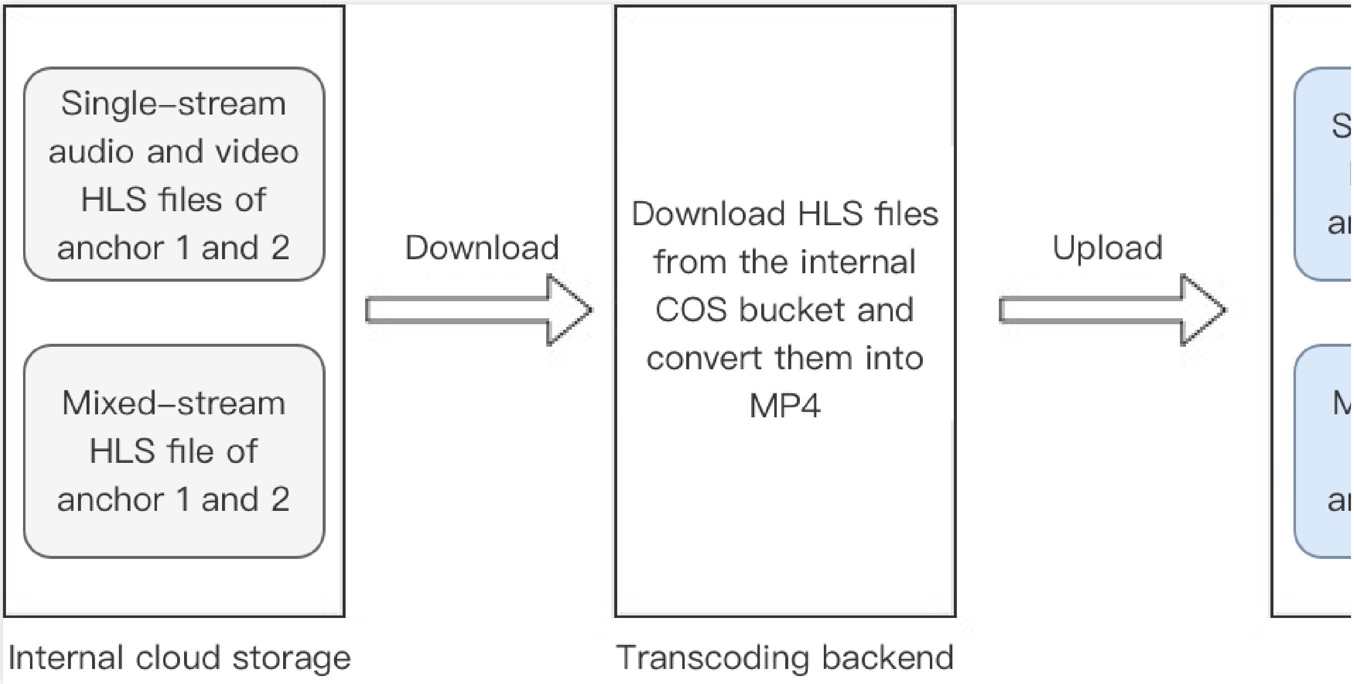


Recording to VOD

To record to VOD, specify the `CloudVod` parameter in `StorageParams` when calling the `CreateCloudRecording` API. After recording ends, the backend will save the file in MP4 format to VOD using the method you specify and send you a playback URL via a callback. In the single-stream recording mode, there will be a playback URL for each anchor whose stream is recorded; in the mixed-stream recording mode, there will be only one playback URL. When you record to VOD, pay attention to the following:

1. `CloudVod` and `CloudStorage` are mutually exclusive. If you specify both, the recording task will fail to start.
2. If you use `DescribeCloudRecording` to query a recording to VOD task, the `StorageFile` returned will be empty.

The figure below shows the workflow of recording to VOD. "Internal cloud storage" refers to the internal storage of the recording backend.



Script for merging single-stream recording files

We offer a script for merging single-stream audio and video files into MP4 files.

Note:

If two segment files are more than 15 seconds apart, during which no audio or video data is recorded (if the substream is disabled, its data will be ignored), the two segments will be considered to belong to different sections, one being the ending segment of the previous section, and the other the starting segment of the next section.

Section-based merge (`-m 0`)

In this mode, the recording files of each user (`UserId`) are merged by section. One MP4 file is generated for each section.

User-based merge (`-m 1`)

In this mode, the recording files of each user (`UserId`) are merged into one MP4 file. You can use the `-s` option to specify whether to fill in the blanks between sections.

Environment Requirements

Python 3

Centos: sudo yum install python3

Ubuntu: sudo apt-get install python3

Python 3 dependency

```
sortedcontainers: pip3 install sortedcontainers
```

Directions

1. Run the merge script `python3 TRTC_Merge.py [option]` .
2. An MP4 file will be generated in the directory of the recording files.

Example: python3 TRTC_Merge.py -f /xxx/file -m 0

Below is a list of the options:

Parameter	Description

-f	The directory of the recording files to be merged. If there are multiple users (`UserId`), their recording files will be merged separately.
-m	`0` : Section-based merge (default). In this mode, the recording files of each user (`UserId`) are merged by section. Multiple files may be generated for each user. `1` : User-based merge. In this mode, the recording files of each user (`UserId`) are merged into one file.
-s	Whether to delete the blanks between sections in the user-based merge mode. If they are deleted, the files generated will be shorter than the recording duration.
-a	`0` : Primary stream merge (default). The audio of a user (`UserId`) is merged with the user's primary stream, not the substream. `1` : Automatic merge. If a user (`UserId`) has a primary stream, the user's audio is merged with the primary stream; if not, it is merged with the substream. `2` : Substream merge. The audio of a user (`UserId`) is merged with the user's substream, not the primary stream.
-p	The frame rate (fps) of the output video, which is `15` by default. Value range: 5-120. If you enter a value smaller than `5` , `5` will be used; if you enter a value greater than `120` , `120` will be used.
-r	The resolution of the output video. For example, `-r 640 360` indicates that the resolution of the output video is 640 x 360.

File Naming

Audio-video file: UserId_timestamp_av.mp4

Audio-only file: UserId_timestamp.m4a

Note:

`UserId` is the Base64-encoded ID of a user whose stream is recorded. For details, see the naming of recording files. `timestamp` is the starting time of the first TS segment of a section.

Callback APIs

You can register callbacks by providing an HTTP/HTTPS gateway to receive callbacks. When an on-cloud recording event occurs, the system will send a callback notification to your callback server.

Callback format

Callbacks are sent to your server in the form of HTTP/HTTPS POST requests.

Character encoding: UTF-8

Request: The request body is in JSON format.

Response: HTTP STATUS CODE = 200. The server ignores the content of the response packet. For protocol-friendliness, we recommend adding `JSON:` {"code":0}`` to the response.

Parameter description

The header of a callback contains the following fields.

Field	Description
Content-Type	application/json
Sign	The signature.
SdkAppId	The application ID.

The body of a callback contains the following fields.

Field	Type	Description
EventGroupId	Number	The event group ID, which is `3` for on-cloud recording.
EventType	Number	The event type.
CallbackTs	Number	The Unix timestamp (ms) when the callback was sent to your server.
EventInfo	JSON Object	The event information.

Event types

Field	Type	Description
EVENT_TYPE_CLOUD_RECORDING_RECORDER_START	301	On-cloud recording - The recording module was started.
EVENT_TYPE_CLOUD_RECORDING_RECORDER_STOP	302	On-cloud recording - The recording module was stopped.
EVENT_TYPE_CLOUD_RECORDING_UPLOAD_START	303	On-cloud recording - The upload module was started.
EVENT_TYPE_CLOUD_RECORDING_FILE_INFO	304	On-cloud recording - The first M3U8 file was generated and uploaded successfully.

EVENT_TYPE_CLOUD_RECORDING_UPLOAD_STOP	305	On-cloud recording - The files are uploaded.
EVENT_TYPE_CLOUD_RECORDING_FAILOVER	306	On-cloud recording - The recording task was migrated.
EVENT_TYPE_CLOUD_RECORDING_FILE_SLICE	307	On-cloud recording - An M3U8 file was generated (the first TS segment was generated).
EVENT_TYPE_CLOUD_RECORDING_UPLOAD_ERROR	308	On-cloud recording - The upload module encountered an error.
EVENT_TYPE_CLOUD_RECORDING_DOWNLOAD_IMAGE_ERROR	309	On-cloud recording - An error occurred when downloading the image decoding file.
EVENT_TYPE_CLOUD_RECORDING_MP4_STOP	310	On-cloud recording - An MP4 recording task is finished. The callback includes the name and other details of the MP4 file generated.
EVENT_TYPE_CLOUD_RECORDING_VOD_COMMIT	311	On-cloud recording - The recording files were uploaded to VOD.
EVENT_TYPE_CLOUD_RECORDING_VOD_STOP	312	On-cloud recording - A recording to VOD task is finished.

Event information

Field	Type	Description
RoomId	String/Number	The room ID, which must be the same data type as the room ID on the client.
EventTs	Number	The Unix timestamp (seconds) when the event occurred.
UserId	String	The user ID of the recording robot.
TaskId	String	The ID of a recording task.

Payload	JsonObject	This parameter is defined differently for different event types.

If the event type is `301` (EVENT_TYPE_CLOUD_RECORDING_RECORDER_START):

Field	Type	Description
Status	Number	`0` : The recording module was started successfully. `1` : Failed to start the recording module.

```
{
        "EventGroupId": 3,
        "EventType":    301,
        "CallbackTs":   1622186275913,
        "EventInfo":    {
                "RoomId":       "xx",
                "EventTs":      "1622186275",
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0
                }
        }
}
```


If the event type is `302` (EVENT_TYPE_CLOUD_RECORDING_RECORDER_STOP):

Field	Type	Description
LeaveCode	Number	`0` : An API was called to stop the recording module. `1` : The recording robot was removed from the room. `2` : You closed the room. `3` : The server removed the recording robot from the room. `4` : The server closed the room. `99` : There were no anchors in the room, and the recording robot left after the specified time period elapsed. `100` : The room timed out. `101` : The recording robot was removed due to repeat entry by the same user.


```
{
        "EventGroupId": 3,
        "EventType":    302,
        "CallbackTs":   1622186354806,
        "EventInfo":    {
                "RoomId":       "xx",
                "EventTs":      "1622186354",
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "LeaveCode":    0
```

```
                    }
            }
    }
```

If the event type is `303` (EVENT_TYPE_CLOUD_RECORDING_UPLOAD_START):

Field	Type	Description
Status	Number	`0` : The upload module was started successfully. `1` : Failed to start the upload module.

```
{
        "EventGroupId": 3,
        "EventType":    303,
        "CallbackTs":   1622191965320,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191965,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0
                }
        }
}
```

If the event type is `304` (EVENT_TYPE_CLOUD_RECORDING_FILE_INFO):

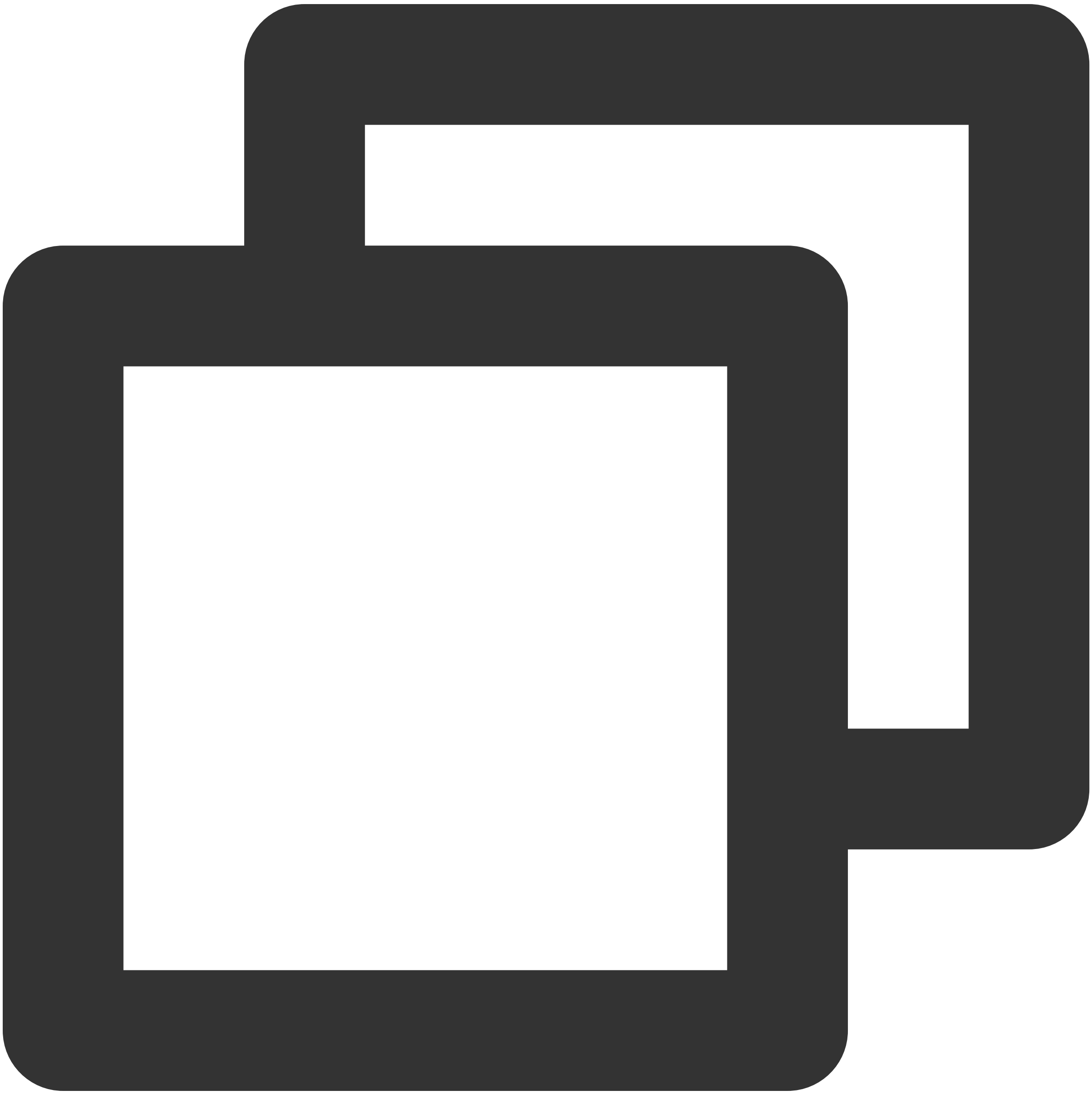
Field	Type	Description
FileList	String	The name of the M3U8 file generated.

```
{
        "EventGroupId": 3,
        "EventType":    304,
        "CallbackTs":   1622191965350,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191965,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "FileList":     "xx.m3u8"
```

```
                    }
            }
    }
```

If the event type is `305` (EVENT_TYPE_CLOUD_RECORDING_UPLOAD_STOP):

Field	Type	Description
Status	Number	`0` : The recording task is finished and all the files were uploaded to the specified cloud storage service. `1` : The recording task is finished, but at least one file is still on the server or in backup storage. `2` : The files on the server or in backup storage were uploaded to the specified cloud storage service.

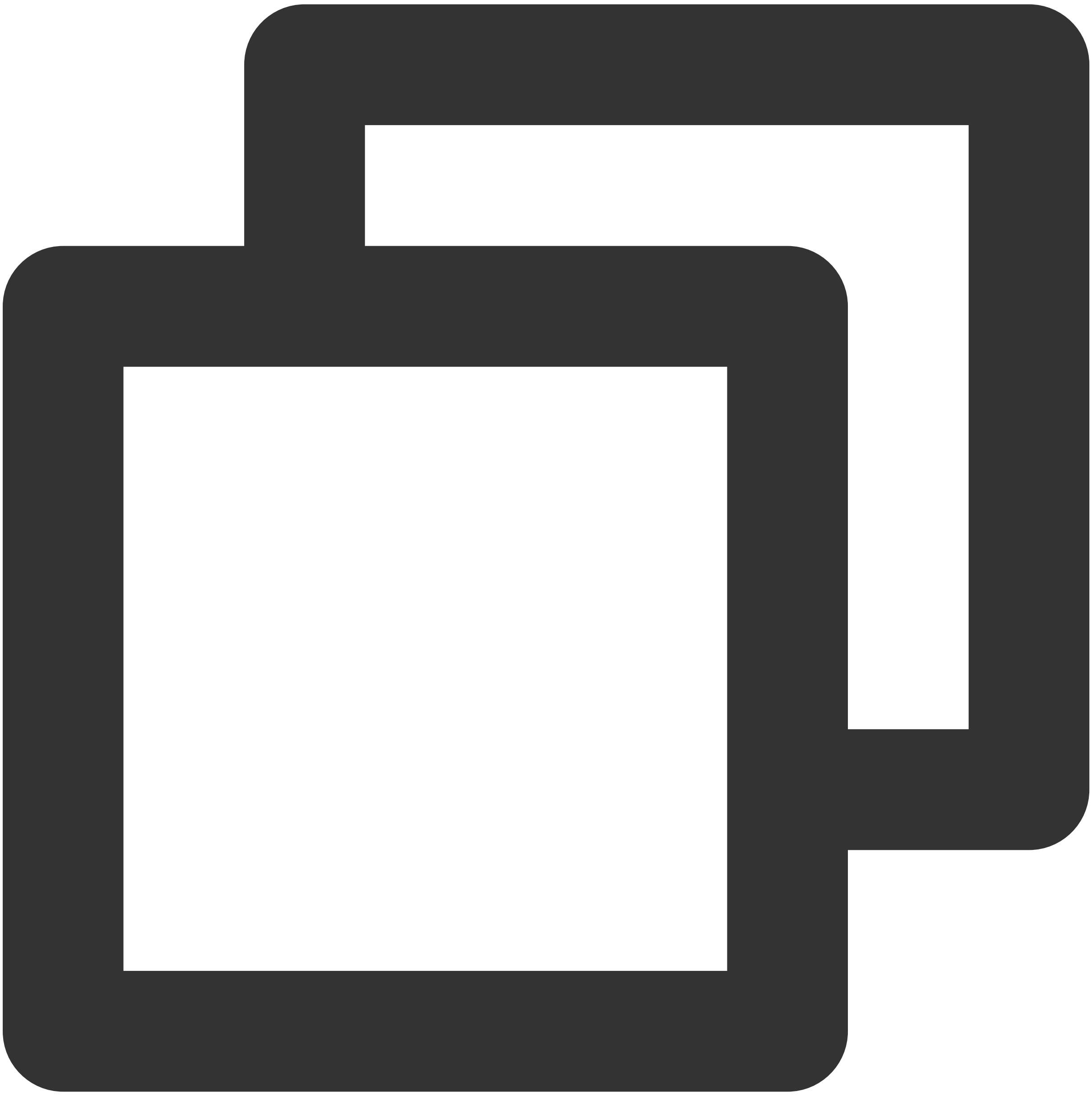


```
{
        "EventGroupId": 3,
        "EventType":    305,
        "CallbackTs":   1622191989674,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191989,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0
```

```
                        }
            }
  }
```

If the event type is `306` (EVENT_TYPE_CLOUD_RECORDING_FAILOVER):

Field	Type	Description
Status	Number	`0` : The migration was completed.




```
{
        "EventGroupId": 3,
        "EventType":    306,
        "CallbackTs":   1622191989674,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191989,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0
                }
        }
}
```

If the event type is `307` (EVENT_TYPE_CLOUD_RECORDING_FILE_SLICE):

Field	Type	Description
FileName	String	The name of the M3U8 file.
UserId	String	The ID of the user whose streams were recorded.
TrackType	String	Valid values: `audio` , `video` , `audio_video` .
BeginTimeStamp	Number	The Unix timestamp (milliseconds) on the server when recording started.

```
{
        "EventGroupId": 3,
        "EventType":    307,
        "CallbackTs":   1622186289148,
        "EventInfo":    {
                "RoomId":       "xx",
                "EventTs":      "1622186289",
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "FileName":     "xx.m3u8",
```



```
                                "UserId":       "xx",
                                "TrackType":    "audio",
                                "BeginTimeStamp":       1622186279144
                      }
            }
}
```

If the event type is `308` (EVENT_TYPE_CLOUD_RECORDING_UPLOAD_ERROR):

Field	Type	Description
Code	String	The error code returned by the cloud storage service.
Message	String	The error message returned by the cloud storage service.


```
{
  "Code": "InvalidParameter",
  "Message": "AccessKey invalid"
}

{
        "EventGroupId": 3,
        "EventType":    308,
        "CallbackTs":   1622191989674,
        "EventInfo":    {
                "RoomId":       "20015",
```

```
                    "EventTs":       1622191989,
                    "UserId":        "xx",
                    "TaskId":        "xx",
                    "Payload":       {
                            "Code":        "xx",
                            "Message":     "xx"
                    }
            }
    }
```


If the event type is `309` (EVENT_TYPE_CLOUD_RECORDING_DOWNLOAD_IMAGE_ERROR):

Field	Type	Description
Url	String	The download URL.

```
{
        "EventGroupId": 3,
        "EventType":    309,
        "CallbackTs":   1622191989674,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191989,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Url":       "http://xx",
```

```
                    }
            }
    }
```

If the event type is `310` (EVENT_TYPE_CLOUD_RECORDING_MP4_STOP):

Field	Type	Description
Status	Number	`0` : The MP4 recording task is finished and all the files were uploaded to the specified cloud storage service. `1` : The MP4 recording task is finished, but at least one file is still on the server or in backup storage. `2` : The MP4 recording task stopped due to an error (probably because the system failed to download the HLS files from COS).
FileList	Array	The names of the MP4 files generated.
FileMessage	Array	The information of the MP4 files generated.
FileName	String	The filename.
UserId	String	The user ID. In the mixed-stream recording mode, this field is empty.
TrackType	String	Valid values: `audio` , `video` , `audio_video` .
MediaId	String	Valid values: `main` , `aux` .
StartTimeStamp	Number	The Unix timestamp (milliseconds) when the MP4 file started.
EndTimeStamp	Number	The Unix timestamp (milliseconds) when the MP4 file ended.

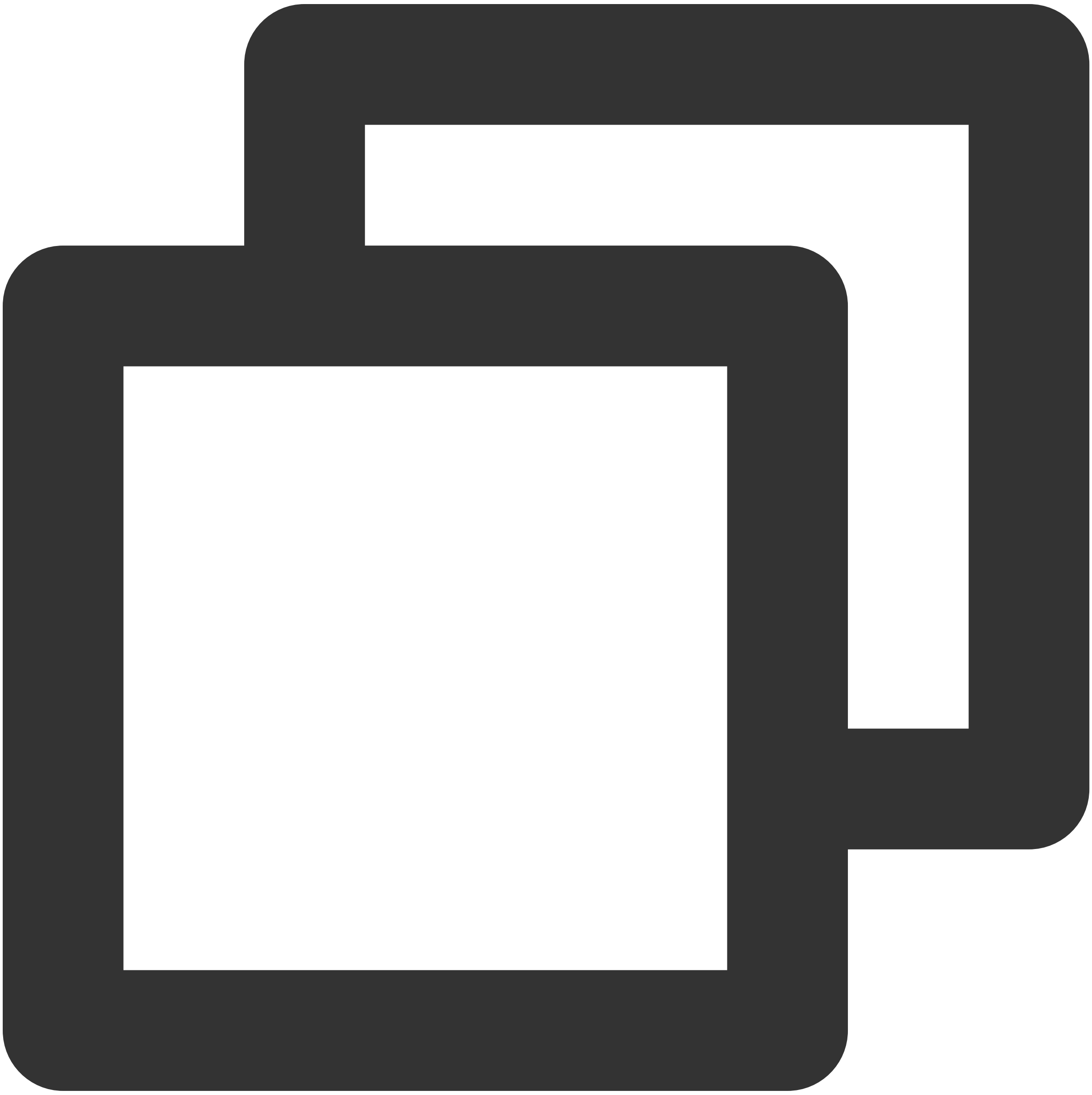
```
{
        "EventGroupId": 3,
        "EventType":    310,
        "CallbackTs":   1622191989674,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191989,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0,
```

```
                        "FileList": ["xxxx1.mp4", "xxxx2.mp4"],
                        "FileMessage": [
                                {
                                        "FileName": "xxxx1.mp4",
                                        "UserId": "xxxx",
                                        "TrackType": "audio_video",
                                        "MediaId":      "main",
                                        "StartTimeStamp": 1622186279145,
                                        "EndTimeStamp": 1622186282145
                                },
                                {
                                        "FileName": "xxxx2.mp4",
                                        "UserId": "xxxx",
                                        "TrackType": "audio_video",
                                        "MediaId":      "main",
                                        "StartTimeStamp": 1622186279153,
                                        "EndTimeStamp": 1622186282153
                                }
                        ]
                }
        }
}
```

If the event type is `311` (EVENT_TYPE_CLOUD_RECORDING_VOD_COMMIT):

Field	Type	Description
Status	Number	`0` : The recording file was successfully uploaded to VOD. `1` : The recording file is still on the server or in backup storage. `2` : An error occurred when uploading the recording file to VOD.
UserId	String	The user ID. In the mixed-stream recording mode, this field is empty.
TrackType	String	Valid values: `audio` , `video` , `audio_video` .
MediaId	String	Valid values: `main` , `aux` .
FileId	String	The ID of the recording file in VOD.
VideoUrl	String	The playback URL of the recording file in VOD.
CacheFile	String	The name of the MP4 file before it was uploaded to VOD.
Errmsg	String	The error message. This field is not empty if `status` is not `0` .

A callback for successful upload:




```
{
        "EventGroupId": 3,
        "EventType":    311,
        "CallbackTs":   1622191965320,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191965,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0,
```

```
                          "TencentVod":   {
                              "UserId":        "xx",
                              "TrackType":     "audio_video",
                              "MediaId":       "main",
                              "FileId":        "xxxx",
                              "VideoUrl":      "http://xxxx"
                          }
                  }
          }
  }
```

A callback for failed upload:

```
{
        "EventGroupId": 3,
        "EventType":    311,
        "CallbackTs":   1622191965320,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191965,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       1,
```



```
                                "Errmsg":       "xxx",
                                "TencentVod":   {
                                    "UserId":       "123",
                                    "TrackType":    "audio_video",
                                    "CacheFile":    "xxx.mp4"
                                }
                        }
                }
}
```

If the event type is `312` (EVENT_TYPE_CLOUD_RECORDING_VOD_STOP):

Field	Type	Description
Status	Number	`0` : The recording to VOD task ended normally. `1` : The recording to VOD task ended due to an error.

```
{
        "EventGroupId": 3,
        "EventType":    312,
        "CallbackTs":   1622191965320,
        "EventInfo":    {
                "RoomId":       "20015",
                "EventTs":      1622191965,
                "UserId":       "xx",
                "TaskId":       "xx",
                "Payload":      {
                        "Status":       0
```

```
                    }
            }
    }
```

Best Practices

To ensure the high availability of the recording service, we recommend the following practices when you use the RESTful APIs:

1. Pay attention to the HTTP response after you call `CreateCloudRecording` . If your request fails, fix the problem according to the status code and try again.

The status code consists of two parts, for example `InvalidParameter.SdkAppId` .

`InternalError.xxxxx` indicates that a server error occurred. You can retry until the request succeeds and `TaskId` is returned. We recommend you use the exponential backup algorithm for retry. For example, you can wait for three seconds for the first retry, six seconds for the second, 12 seconds for the third, and so on.

`InvalidParameter.xxxxx` indicates that a parameter value entered was invalid. Please check the parameter.

`FailedOperation.RestrictedConcurrency` indicates that you reached the maximum number (100 by default) of ongoing recording tasks allowed. To raise the limit, please contact technical support.

2. The `UserId` and `UserSig` you pass in when calling `CreateCloudRecording` are for the recording robot. Please make sure that they are different from those of other users in the room. In addition, the room joined by users from the TRTC client must be of the same type as the room you specify when calling the API. For example, if the room created in the TRTC SDK is a string, the room specified for on-cloud recording must also be a string.

3. You can obtain the information of a recording file in the following ways.

Call `DescribeCloudRecording` 15 seconds after a `CreateCloudRecording` request succeeds. If the task status is idle, it indicates that no audio or video data is available for recording. Please check whether there are anchors publishing data in the room.

After a `CreateCloudRecording` request succeeds, if there are anchors publishing data in the room, you can splice the names of the recording files according to the naming rules.

If you have registered on-cloud recording callbacks, the information of recording files will be sent to your server via callbacks.

You can specify a COS bucket to save recording files when calling the `CreateCloudRecording` API. After a recording task ends, you can find the recording files in the COS bucket you specify.

4. Make sure the validity period of the recording user's `UserSig` is longer than the duration of the recording task. This is to avoid cases where the high availability scheme fails to resume a recording task after a disconnection because the `UserSig` has expired.

Custom Capturing and Rendering Android, iOS, Windows, and macOS

Last updated：2023-10-08 15:41:41

This document describes how to use the TRTC SDK to implement custom video capturing and rendering.

Custom Video Capturing

The custom video capturing feature of the TRTC SDK can be used in two steps: enabling the feature and sending video frames to the SDK. For detailed directions of specific APIs, see below. We also provide API examples for different platforms:

Android

iOS

Windows

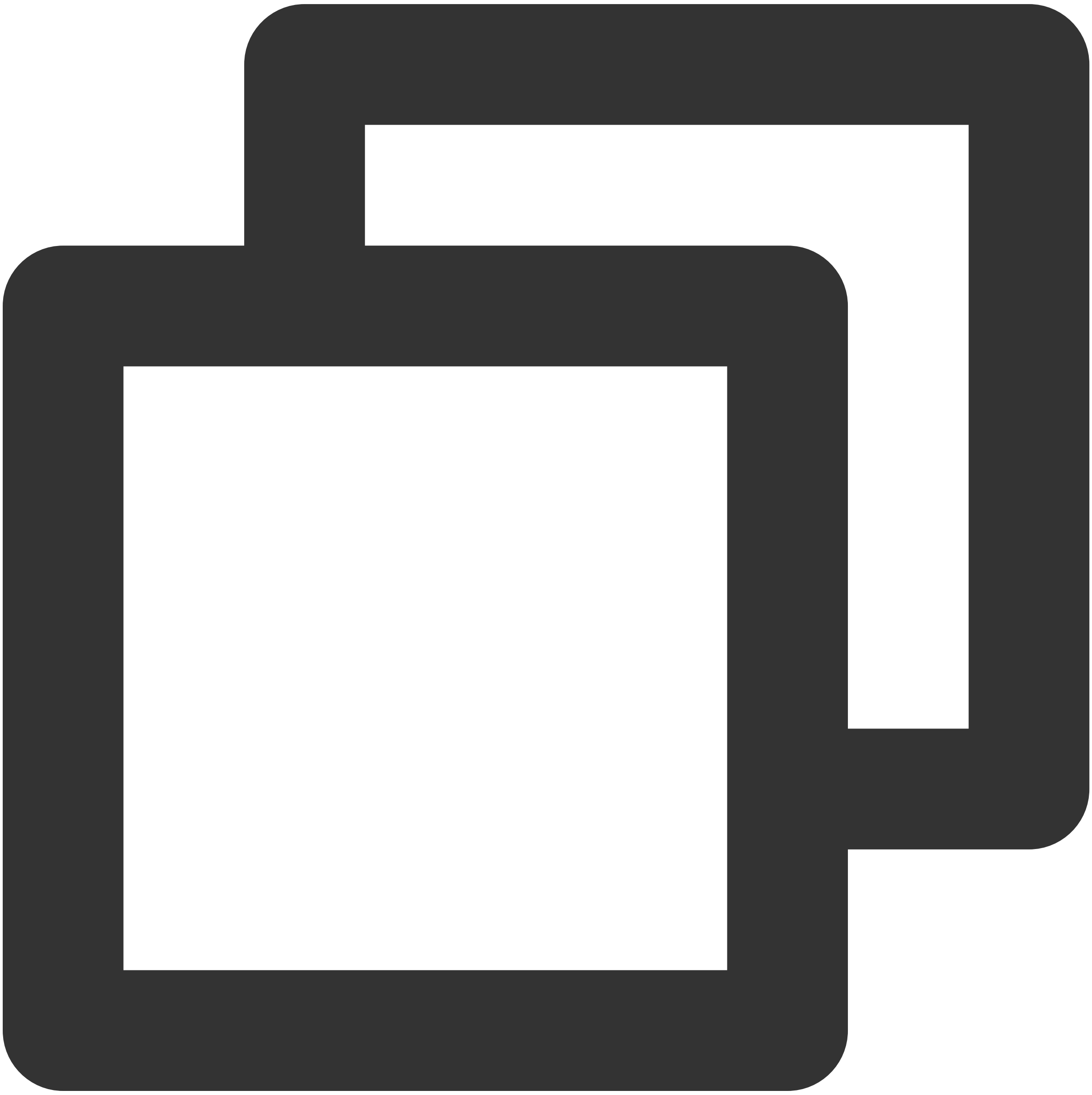
Enabling custom video capturing

To enable the custom video capturing feature of the TRTC SDK, you need to call the `enableCustomVideoCapture` API of `TRTCCloud`. Then, the TRTC SDK's camera capturing and image processing logic will be skipped, and only its encoding and transfer capabilities will be retained. Below is the sample code:

Android

iOS&Mac

Windows



```
TRTCCloud mTRTCCloud = TRTCCloud.shareInstance();
mTRTCCloud.enableCustomVideoCapture(TRTCCloudDef.TRTC_VIDEO_STREAM_TYPE_BIG, true);
```

```
self.trtcCloud = [TRTCCloud sharedInstance];
[self.trtcCloud enableCustomVideoCapture:TRTCVideoStreamTypeBig enable:YES];
```

```
liteav::ITRTCCloud* trtc_cloud = liteav::ITRTCCloud::getTRTCShareInstance();
trtc_cloud->enableCustomVideoCapture(TRTCVideoStreamType::TRTCVideoStreamTypeBig, t
```

Sending custom video frames

Then, you can use the `sendCustomVideoData` API of `TRTCCloud` to populate the TRTC SDK with your own video data. Below is the sample code:

explain

In order to avoid performance loss, there are different format requirements for the video data input to the TRTC SDK on different platforms. For more information, see LiteAVSDK Overview.

Android

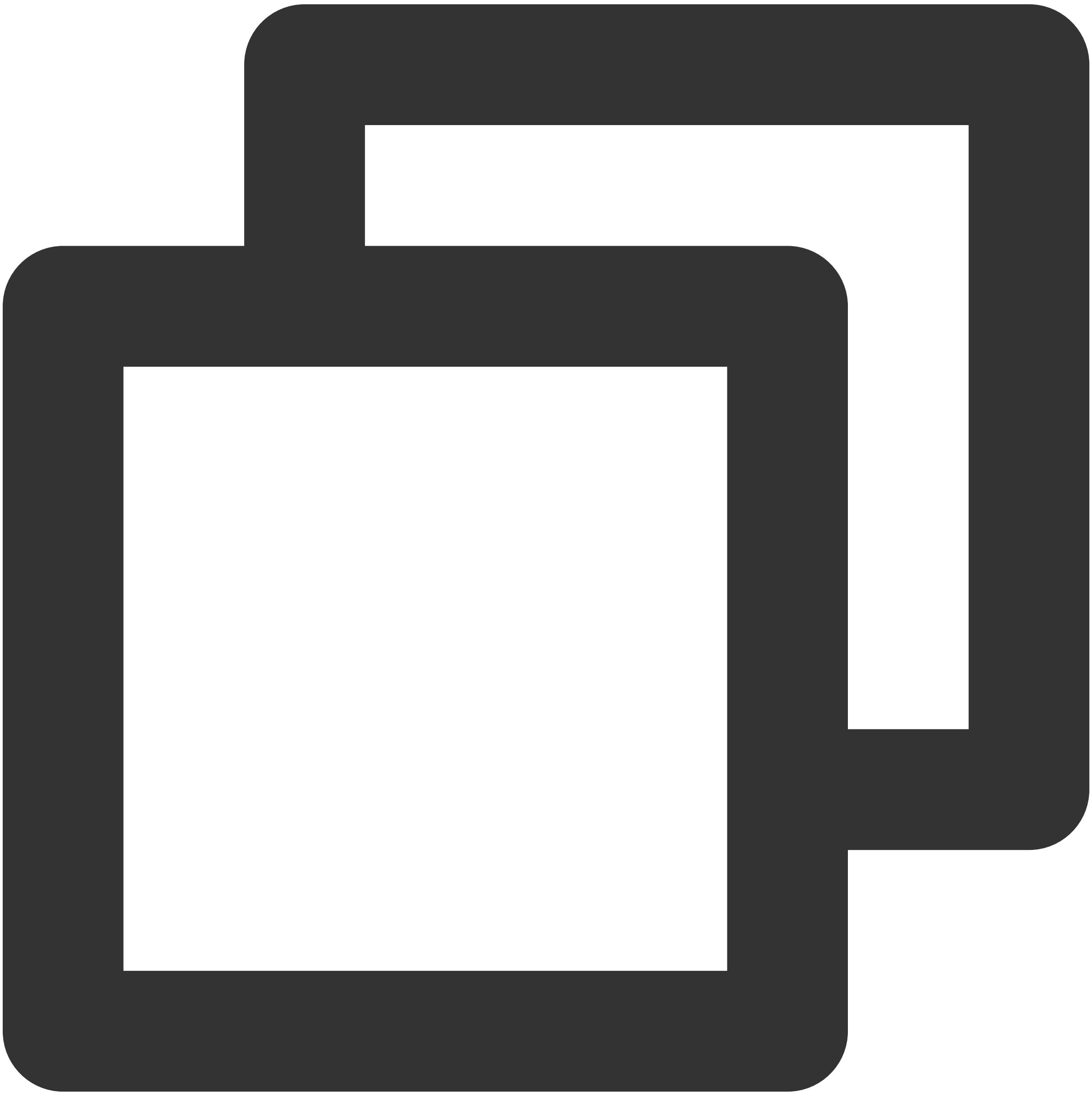
iOS&Mac

Windows

```
// Two schemes are available for Android: Texture (recommended) and Buffer. Texture
TRTCCloudDef.TRTCVideoFrame videoFrame = new TRTCCloudDef.TRTCVideoFrame();
videoFrame.texture = new TRTCCloudDef.TRTCTexture();
videoFrame.texture.textureId = textureId;
```



```
videoFrame.texture.eglContext14 = eglContext;
videoFrame.width = width;
videoFrame.height = height;
videoFrame.timestamp = timestamp;
videoFrame.pixelFormat = TRTCCloudDef.TRTC_VIDEO_PIXEL_FORMAT_Texture_2D;
videoFrame.bufferType = TRTCCloudDef.TRTC_VIDEO_BUFFER_TYPE_TEXTURE;
mTRTCCloud.sendCustomVideoData(TRTCCloudDef.TRTC_VIDEO_STREAM_TYPE_BIG, videoFrame)
```



```
// On iOS and macOS, the video captured by the camera is in NV12 format. The native
TRTCVideoFrame *videoFrame = [[TRTCVideoFrame alloc] init];
```

```
videoFrame.pixelFormat = TRTCVideoPixelFormat_NV12;
videoFrame.bufferType = TRTCVideoBufferType_PixelBuffer;
videoFrame.pixelBuffer = imageBuffer;
videoFrame.timestamp = timeStamp;

[[TRTCCloud sharedInstance] sendCustomVideoData:TRTCVideoStreamTypeBig frame:videoF
```

```
// Only the Buffer scheme is available for Windows currently and is recommended for
liteav::TRTCVideoFrame frame;
frame.timestamp = getTRTCShareInstance()->generateCustomPTS();
```

```
frame.videoFormat = liteav::TRTCVideoPixelFormat_I420;
frame.bufferType = liteav::TRTCVideoBufferType_Buffer;
frame.length = buffer_size;
frame.data = array.data();
frame.width = YUV_WIDTH;
frame.height = YUV_HEIGHT;
getTRTCShareInstance()->sendCustomVideoData(&frame);
```

Custom Video Rendering

Custom rendering is mainly divided into rendering of the local video and rendering of the remote video. You can set the callback for local/remote custom rendering, and the TRTC SDK will pass the corresponding video frames (`TRTCVideoFrame`) through the callback function `onRenderVideoFrame` . Then, you can customize the rendering of the received video frames. This process requires certain knowledge of OpenGL. We also provide API examples for different platforms:

Android:

iOS

Windows

Setting the local video rendering callback

Android

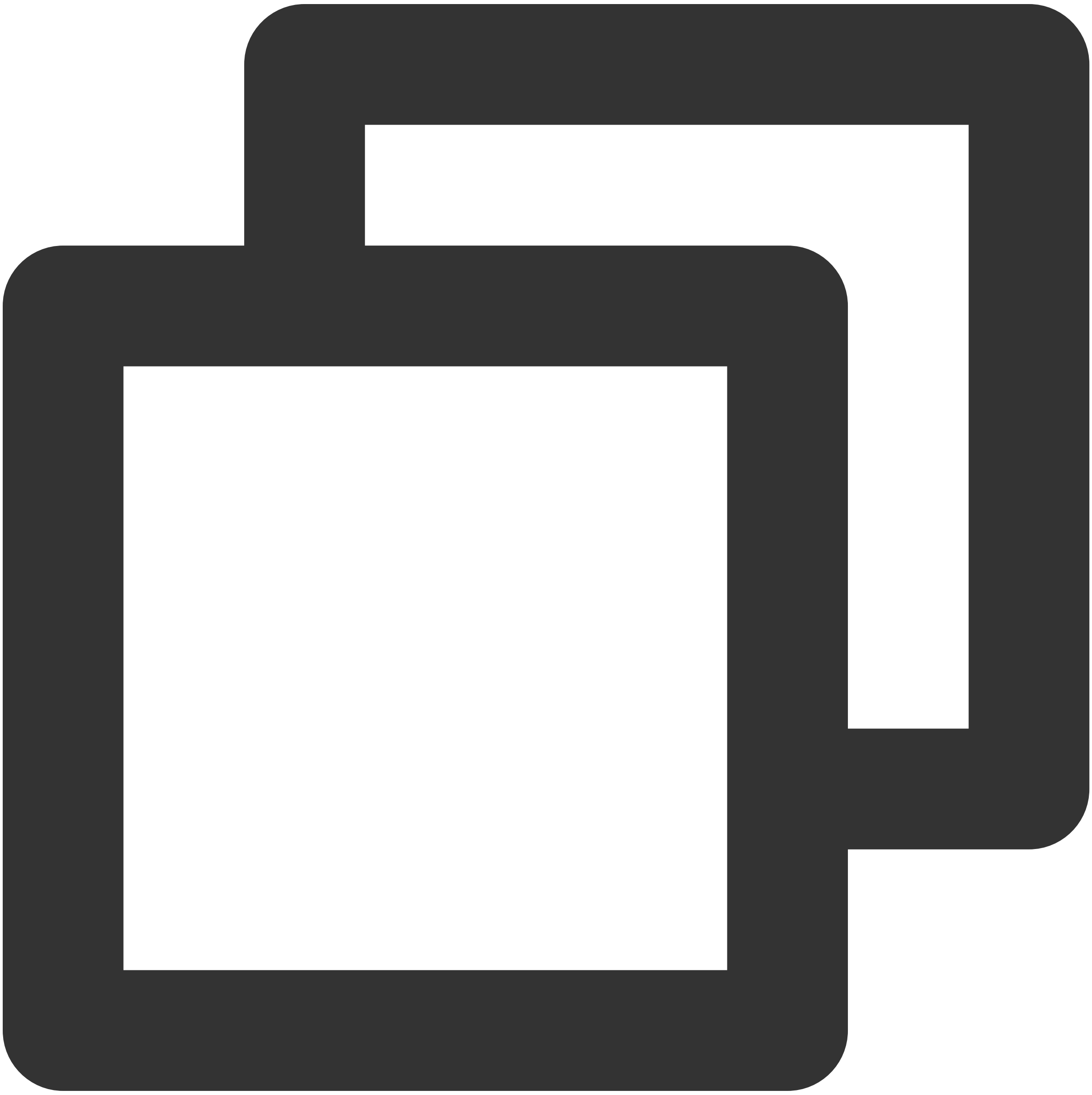
iOS&Mac

Windows


```
mTRTCCloud.setLocalVideoRenderListener(TRTCCloudDef.TRTC_VIDEO_PIXEL_FORMAT_Texture
    @Override
    public void onRenderVideoFrame(String suserId int streamType, TRTCCloudDef.TRTC
        // For more information, see the custom rendering tool class `com.tencent.t
    }
});
```

```
self.trtcCloud = [TRTCCloud sharedInstance];
[self.trtcCloud setLocalVideoRenderDelegate:self pixelFormat:TRTCVideoPixelFormat_N
```

```
}
```



````
```
// For specific implementation, see `test_custom_render.cpp` in `TRTC-API-Example-Q
void TestCustomRender::onRenderVideoFrame(
    const char* userId,
    liteav::TRTCVideoStreamType streamType,
    liteav::TRTCVideoFrame* frame) {
  if (gl_yuv_widget_ == nullptr) {
    return;
  }

  if (streamType == liteav::TRTCVideoStreamType::TRTCVideoStreamTypeBig) {
````
````

```
    // Adjust the rendering window
    emit renderViewSize(frame->width, frame->height);
    // Draw video frames
    gl_yuv_widget_->slotShowYuv(reinterpret_cast<uchar*>(frame->data),
                                frame->width, frame->height);
  }
}
```
```

### Setting the rendering callback of remote video

Android

iOS&Mac

Windows
```

```
mTRTCCloud.setRemoteVideoRenderListener(userId, TRTCCloudDef.TRTC_VIDEO_PIXEL_FORMA
    @Override
    public void onRenderVideoFrame(String userId, int streamType, TRTCCloudDef.TRTC
         // For more information, see the custom rendering tool class `com.tencent.
    }
});
```



```
- (void)onRenderVideoFrame:(TRTCVideoFrame *)frame
                    userId:(NSString *)userId
                streamType:(TRTCVideoStreamType)streamType
{
    // If `userId` is `nil`, the image rendered is the local image; otherwise it is
    CFRetain(frame.pixelBuffer);
    __weak __typeof(self) weakSelf = self;
    dispatch_async(dispatch_get_main_queue(), ^{
        TestRenderVideoFrame *strongSelf = weakSelf;
        UIImageView* videoView = nil;
        if (userId) {
```

```
            videoView = [strongSelf.userVideoViews objectForKey:userId];
        }
        else {
            videoView = strongSelf.localVideoView;
        }
        videoView.image = [UIImage imageWithCIImage:[CIImage imageWithCVImageBuffer
        videoView.contentMode = UIViewContentModeScaleAspectFit;
        CFRelease(frame.pixelBuffer);
    });
}
```

````
```
// For specific implementation, see `test_custom_render.cpp` in `TRTC-API-Example-Q
void TestCustomRender::onRenderVideoFrame(
    const char* userId,
    liteav::TRTCVideoStreamType streamType,
    liteav::TRTCVideoFrame* frame) {
  if (gl_yuv_widget_ == nullptr) {
    return;
  }

  if (streamType == liteav::TRTCVideoStreamType::TRTCVideoStreamTypeBig) {
    // Adjust the rendering window
    emit renderViewSize(frame->width, frame->height);
    // Draw video frames
    gl_yuv_widget_->slotShowYuv(reinterpret_cast<uchar*>(frame->data),
                                frame->width, frame->height);
  }
}
```
````

Web

Last updated：2023-10-08 15:55:32

Function Description

This article mainly introduces the advanced usage of custom capture and custom rendering.

Custom Capture

By default, `trtc.startLocalVideo()` and `trtc.startLocalAudio()` enable camera and microphone capture.

If you need to customize the capture, you can specify the `option.videoTrack` / `option.audioTrack` parameter of the `trtc.startLocalVideo()` / `trtc.startLocalAudio()` method.

There are usually several ways to obtain `audioTrack` and `videoTrack` :

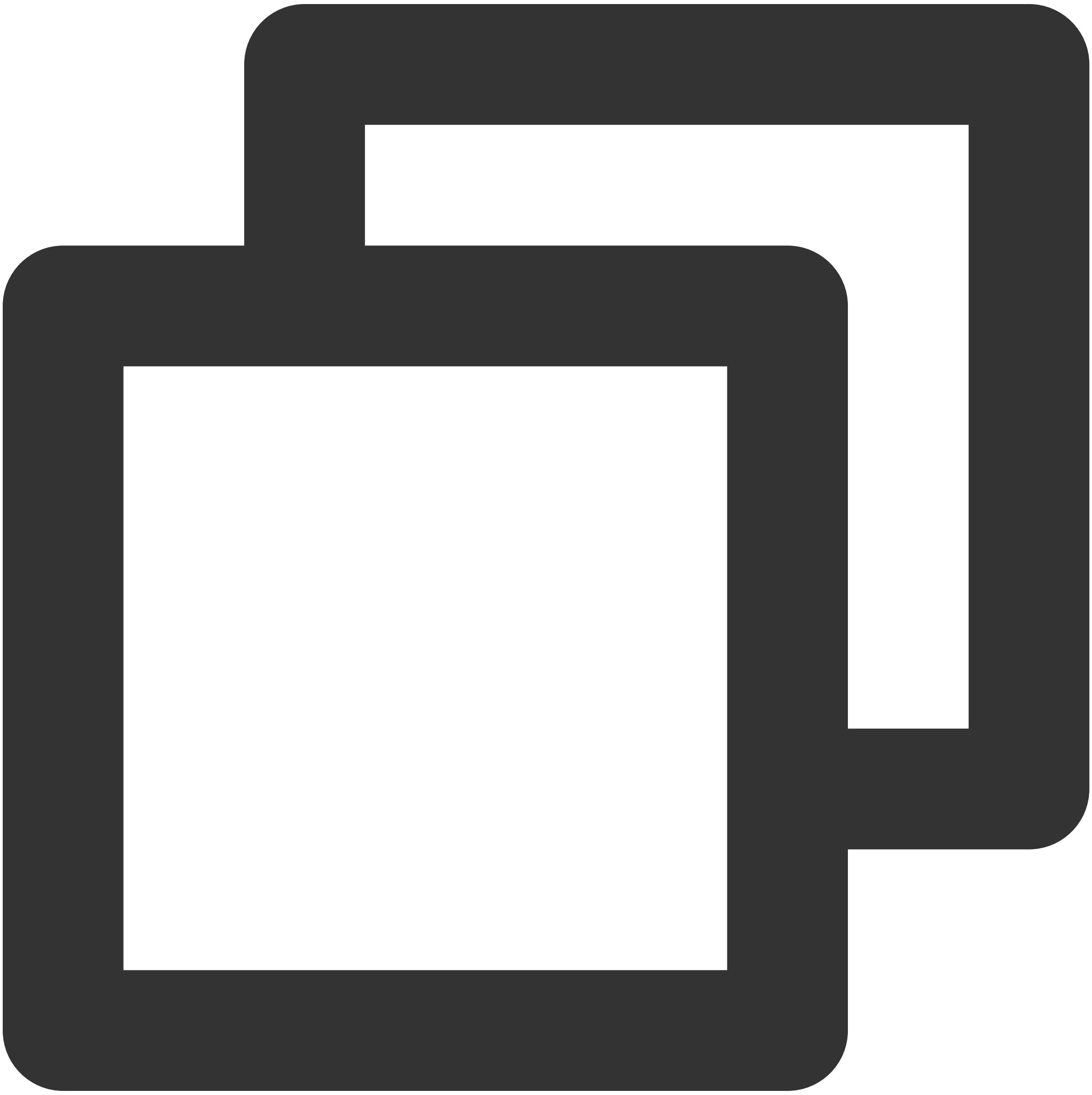
Use getUserMedia to capture the camera and microphone.

Use getDisplayMedia to capture screen sharing.

Use `videoElement.captureStream` to capture the audio and video being played in the video tag.

Use `canvas.captureStream` to capture the animation in the canvas.

Capture the video being played in the video tag



```
// Check if your current browser supports capturing streams from video elements
if (!HTMLVideoElement.prototype.captureStream) {
  console.log('your browser does not support capturing stream from video element');
  return
}
// Get the video tag that is playing video on your page
const video = document.getElementByID('your-video-element-ID');
// Capture the video stream from the playing video
const stream = video.captureStream();
const audioTrack = stream.getAudioTracks()[0];
const videoTrack = stream.getVideoTracks()[0];
```

```
trtc.startLocalVideo({ option:{ videoTrack } });
trtc.startLocalAudio({ option:{ audioTrack } });
```

Capture the animation in the canvas

```
// Check if your current browser supports capturing streams from canvas elements
if (!HTMLCanvasElement.prototype.captureStream) {
  console.log('your browser does not support capturing stream from canvas element')
  return
```

```
}
// Get your canvas tag
const canvas = document.getElementByID('your-canvas-element-ID');

// Capture a 15 fps video stream from the canvas
const fps = 15;
const stream = canvas.captureStream(fps);
const videoTrack = stream.getVideoTracks()[0];

trtc.startLocalVideo({ option:{ videoTrack } });
```

Custom Rendering

By default, when calling `trtc.startLocalVideo(view)` or `trtc.startRemoteVideo(view, streamType, userId)` , you need to pass in the `view` parameter. The SDK will create a `video` tag under the specified element tag to play the video.

If you need to customize the rendering, and do not need the SDK to play the video, you can refer to the following steps:

Do not fill in the `view` parameter or pass in `null` when calling the `startLocalVideo` or `startRemoteVideo` method.

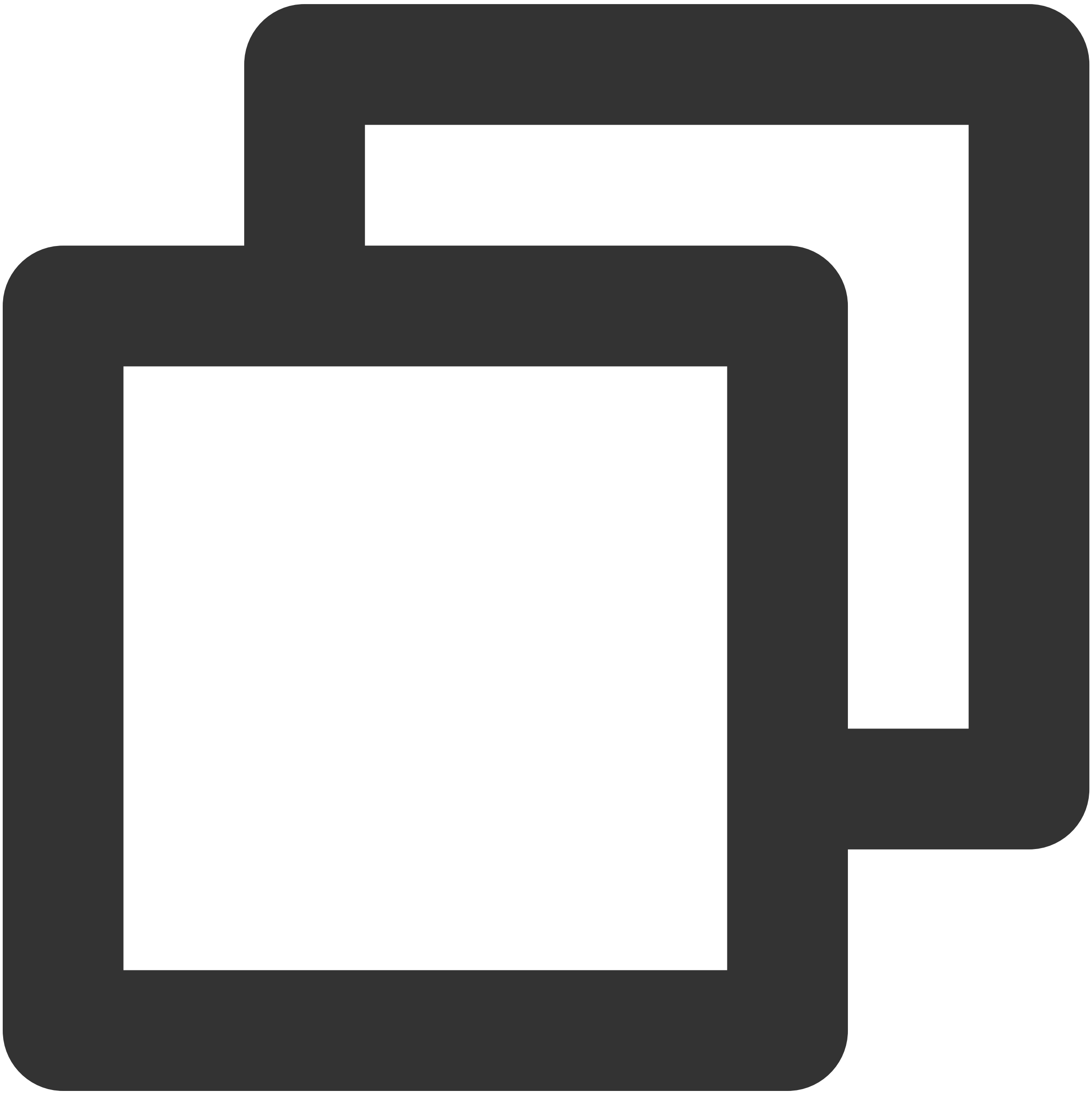
Use the `trtc.getVideoTrack(userId, streamType)` method to obtain the corresponding `videoTrack` .

Use your own player for video rendering.

After using this custom rendering method, the `EVENT.VIDEO_PLAY_STATE_CHANGED` event will not be triggered. You need to listen to the `mute/unmute/ended` events of the video track `MediaStreamTrack` to determine the status of the current video data stream.

For remote video, you also need to listen to the `EVENT.REMOTE_VIDEO_AVAILABLE` and `EVENT.REMOTE_VIDEO_UNAVAILABLE` events to handle the lifecycle of remote video.

Custom rendering of local video



```
await trtc.startLocalVideo();
const videoTrack = trtc.getVideoTrack();

// Use your own player for video rendering
const videoElement = document.getElementById('video-element');
videoElement.srcObject = new MediaStream([videoTrack]);
videoElement.play();
```

Custom rendering of remote video


```
trtc.on(TRTC.EVENT.REMOTE_VIDEO_AVAILABLE, async ({ userId, streamType }) => {
  // Only pull the stream, do not play it
  await trtc.startRemoteVideo({ userId, streamType })
  const videoTrack = trtc.getVideoTrack({ userId, streamType });

  // Use your own player for video rendering
  const videoElement = document.getElementById('remote-video-element');
  videoElement.srcObject = new MediaStream([videoTrack]);
  videoElement.play();
});
```

Flutter

Last updated：2023-12-28 21:30:57

This document mainly introduces how to use TRTC Flutter SDK to implement custom audio raw data acquisition.

Acquiring audio raw data

Flutter TRTC SDK provides two ways to acquire audio raw data:

Native access.

Direct use of Flutter's Dart interface.

Since transferring high-frequency and large audio raw data from Native to Dart layer consumes more performance, we recommend using Native access to acquire audio raw data.

1. Native access

The specific access process and access effect can be experienced using the demo.

1.1 Listen to audio raw data at the Native layer and acquire audio raw data.

java

swift

```
}
```

```
void enableTRTCAudioFrameDelegate() {
    TRTCCloud.sharedInstance(getApplicationContext()).setAudioFrameListener(new Aud
    result.success("");
}

void disableTRTCAudioFrameDelegate() {
    TRTCCloud.sharedInstance(getApplicationContext()).setAudioFrameListener(null);
    result.success("");
}

class AudioFrameListener implements TRTCCloudListener.TRTCAudioFrameListener {
```

```
    @Override
    public void onCapturedAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFrame) {
        // TODO
    }

    @Override
    public void onLocalProcessedAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFra
        // TODO
    }

    @Override
    public void onRemoteUserAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFrame,
        // TODO
    }

    @Override
    public void onMixedPlayAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFrame) {
        // TODO
    }

    @Override
    public void onMixedAllAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFrame) {
        // TODO
    }

    @Override
    public void onVoiceEarMonitorAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFr
        // TODO
    }
}
```

```
}
```

```
let listener = AudioFrameProcessListener()
func enableTRTCAudioFrameDelegate() {
    TRTCCloud.sharedInstance().setAudioFrameDelegate(listener)
    result(nil)
}

func disableTRTCAudioFrameDelegate() {
    TRTCCloud.sharedInstance().setAudioFrameDelegate(nil)
    result(nil)
}
```



```
class AudioFrameProcessListener: NSObject, TRTCAudioFrameDelegate {
    func onCapturedAudioFrame(_ frame: TRTCAudioFrame) {
        //MARK: TODO
    }

    func onLocalProcessedAudioFrame(_ frame: TRTCAudioFrame) {
        // MARK: TODO
    }

    func onRemoteUserAudioFrame(_ frame: TRTCAudioFrame, userId: String) {
        // MARK: TODO
    }

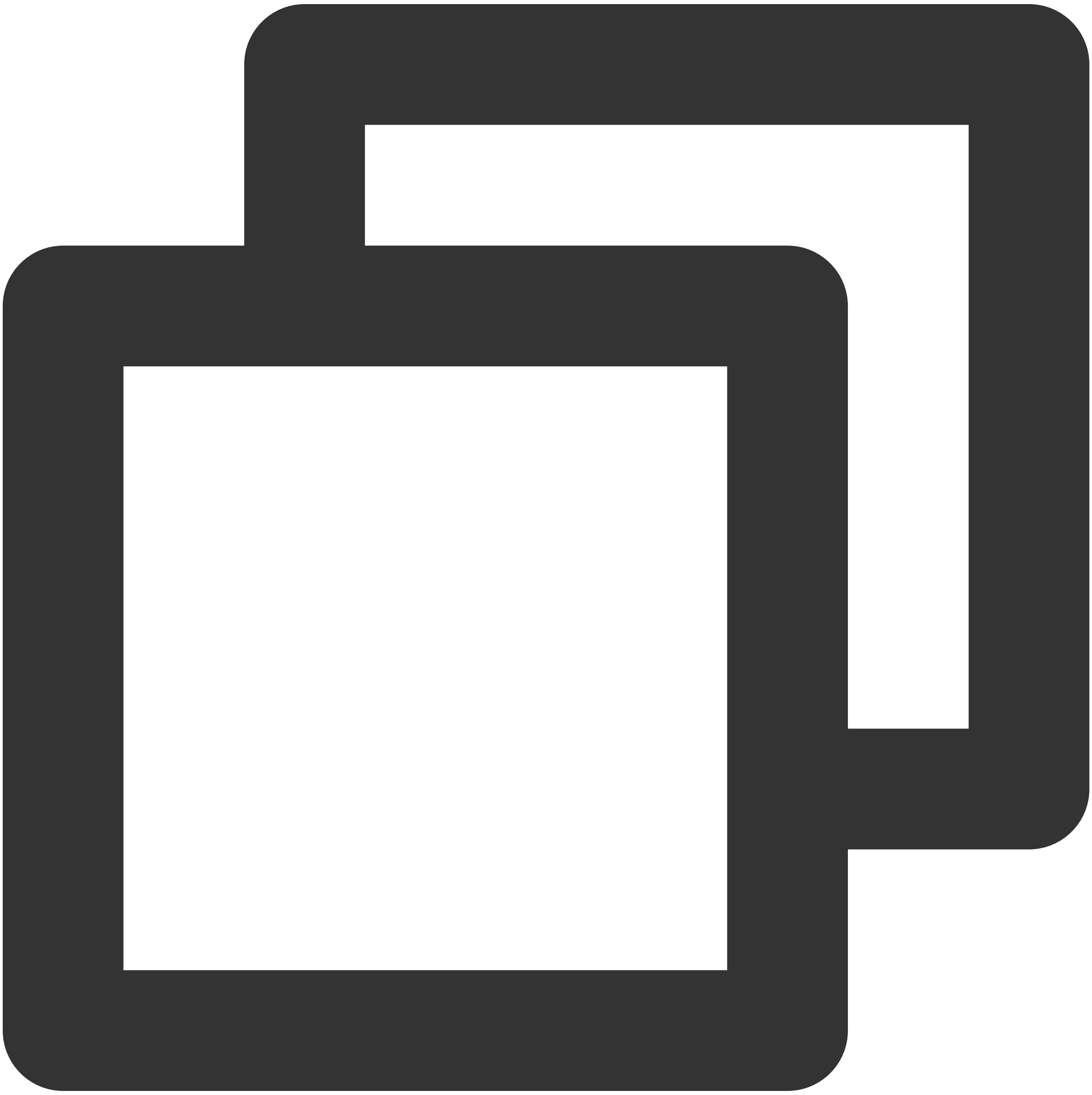
    func onMixedAllAudioFrame(_ frame: TRTCAudioFrame) {
        // MARK: TODO
    }

    func onMixedPlay(_ frame: TRTCAudioFrame) {
        // MARK: TODO
    }

    func onVoiceEarMonitorAudioFrame(_ frame: TRTCAudioFrame) {
        // MARK: TODO
    }
}
```

1.2 Use Method Channel to implement start/stop acquisition of audio raw data.

Step 1: Implement the start/stop interface for acquiring audio raw data at the Dart layer.



```
final channel = MethodChannel('TRCT_FLUTTER_EXAMPLE');

void enableAudioFrame() async {
  await channel.invokeMethod('enableTRTCAudioFrameDelegate');
}

void disableAudioFrame() async {
  await channel.invokeMethod('disableTRTCAudioFrameDelegate');
}
```

Step 2: Implement the start/stop interface for acquiring audio raw data at the Native layer.

java

swift

```
public class MainActivity extends FlutterActivity {
    private static final String channelName = "TRCT_FLUTTER_EXAMPLE";

    private MethodChannel channel;

    @Override
    public void configureFlutterEngine(@NonNull FlutterEngine flutterEngine) {
        super.configureFlutterEngine(flutterEngine);
```

```
        channel = new MethodChannel(flutterEngine.getDartExecutor().getBinaryMessen
        channel.setMethodCallHandler(((call, result) -> {
            switch (call.method) {
                case "enableTRTCAudioFrameDelegate":
                    enableTRTCAudioFrameDelegate();
                    break;
                case "disableTRTCAudioFrameDelegate":
                    disableTRTCAudioFrameDelegate();
                    break;
                default:
                    break;
            }
        }));
    }
}
```

```
@UIApplicationMain
@objc class AppDelegate: FlutterAppDelegate {
    var channel: FlutterMethodChannel?

    override func application(_ application: UIApplication,
                              didFinishLaunchingWithOptions launchOptions: [UIAppli
        GeneratedPluginRegistrant.register(with: self)

        guard let controller = window?.rootViewController as? FlutterViewController
            fatalError("Invalid root view controller")
        }
```

```
        channel = FlutterMethodChannel(name: "TRCT_FLUTTER_EXAMPLE", binaryMessenge
        channel?.setMethodCallHandler({ [weak self] call, result in
            guard let self = self else { return }
            switch (call.method) {
            case "enableTRTCAudioFrameDelegate":
                self.enableTRTCAudioFrameDelegate()
                break
            case "disableTRTCAudioFrameDelegate":
                self.disableTRTCAudioFrameDelegate()
                break
            default:
                break
            }
        })
        return super.application(application, didFinishLaunchingWithOptions: launch
    }
}
```

2. Access to Flutter layer interface

Currently, the Flutter Dart interface only supports the use of the onCapturedAudioFrame interface. The specific usage is as follows:

```
TRTCCloud trtcCloud = (await TRTCCloud.sharedInstance())!;

//  Start acquiring audio raw data
final audioFrameListener = TRTCAudioFrameListener(
  onCapturedAudioFrame: (audioFrame) {
    // TODO
  }
);
trtcCloud.setAudioFrameListener(audioFrameListener);

// Stop acquiring audio raw data
```

```
trtcCloud.setAudioFrameListener(null);
```

Custom Audio Capturing and Playback Android, iOS, Windows, and macOS

Last updated：2023-10-08 15:56:25

This document describes how to use the TRTC SDK to implement custom audio capturing and rendering.

Custom Audio Capturing

The custom audio capturing feature of the TRTC SDK can be used in two steps: enabling the feature and sending audio frames to the SDK. For detailed directions of specific APIs, see below. We also provide API examples for different platforms:

Android

iOS

Windows

Enabling custom audio capturing

To enable the custom audio capturing feature of the TRTC SDK, you need to call the `enableCustomAudioCapture` API of `TRTCCloud`. Below is the sample code:

Android

iOS&Mac

Windows

```
TRTCCloud mTRTCCloud = TRTCCloud.shareInstance();
mTRTCCloud.enableCustomAudioCapture(true);
```

```
self.trtcCloud = [TRTCCloud sharedInstance];
[self.trtcCloud enableCustomAudioCapture:YES];
```

```
liteav::ITRTCCloud* trtc_cloud = liteav::ITRTCCloud::getTRTCShareInstance();
trtc_cloud->enableCustomAudioCapture(true);
```

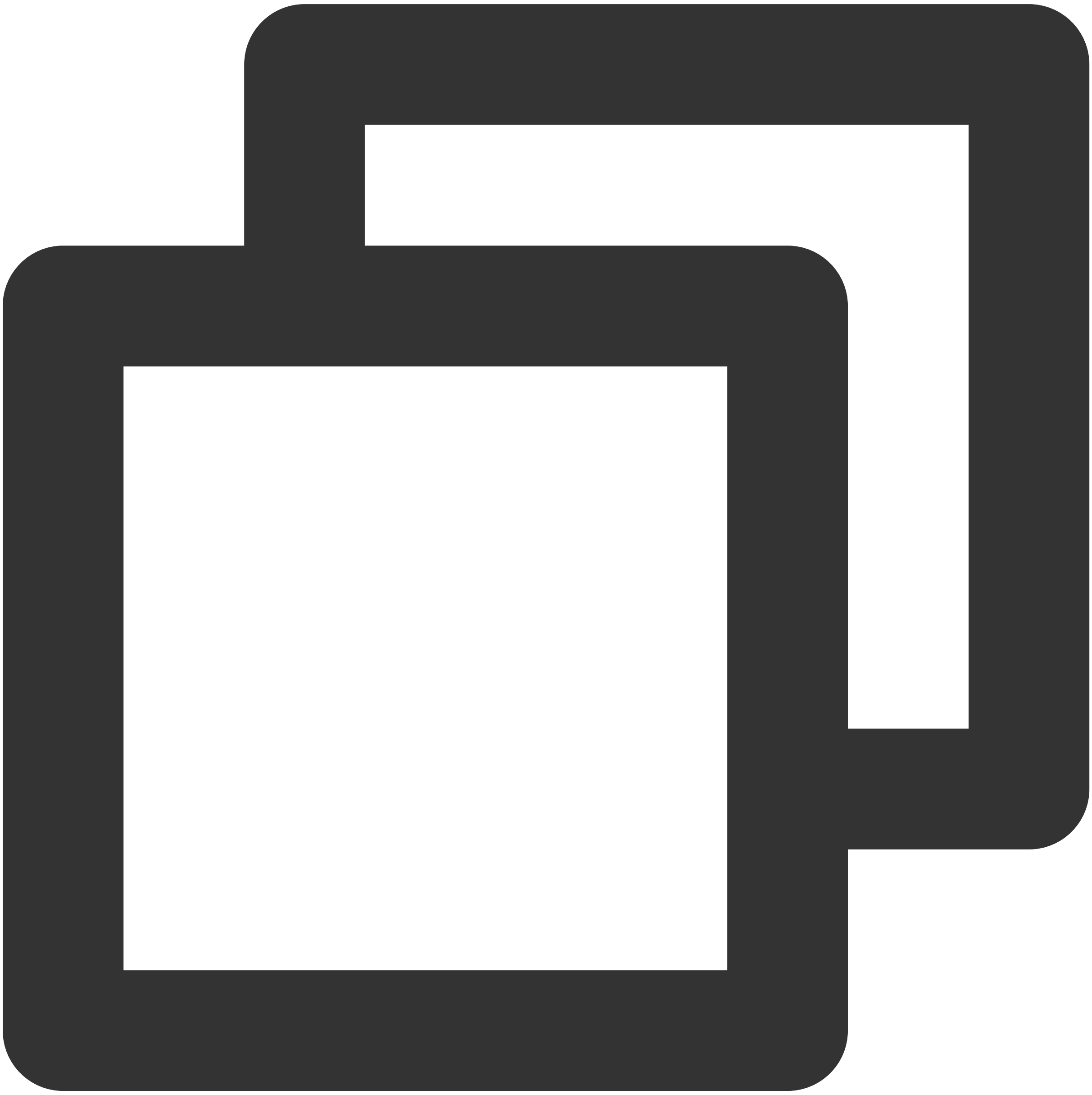
Sending custom audio frames

You can use the `sendCustomAudioData` API of `TRTCCloud` to populate the TRTC SDK with your own audio data. Below is the sample code:

Android

iOS&Mac

Windows



```
TRTCCloudDef.TRTCAudioFrame trtcAudioFrame = new TRTCCloudDef.TRTCAudioFrame();
trtcAudioFrame.data = data;
trtcAudioFrame.sampleRate = sampleRate;
trtcAudioFrame.channel = channel;
trtcAudioFrame.timestamp = timestamp;
mTRTCCloud.sendCustomAudioData(trtcAudioFrame);
```

```
TRTCAudioFrame *audioFrame = [[TRTCAudioFrame alloc] init];
audioFrame.channels = audioChannels;
audioFrame.sampleRate = audioSampleRate;
audioFrame.data = pcmData;

[self.trtcCloud sendCustomAudioData:audioFrame];
```

```
liteav::TRTCAudioFrame frame;
frame.audioFormat = liteav::TRTCAudioFrameFormatPCM;
frame.length = buffer_size;
frame.data = array.data();
frame.sampleRate = 48000;
frame.channel = 1;
getTRTCShareInstance()->sendCustomAudioData(&frame);
```

notice

Using `sendCustomAudioData` may cause AEC to fail.

Getting Raw Audio Data

The audio module is a highly complex module, and the TRTC SDK needs to strictly control the capturing and playback logic of audio devices. In some cases, to get the audio data of a remote user or audio captured by the local mic, you can use the APIs of `TRTCCloud` for different platforms. We also provide API examples for those platforms:

Android:

iOS

Windows

Setting audio callback

Android

iOS&Mac

Windows

```
}
```

```
mTRTCCloud.setAudioFrameListener(new TRTCCloudListener.TRTCAudioFrameListener() {
        @Override
        public void onCapturedRawAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFr

        }

        @Override
        public void onLocalProcessedAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudi

        }
```

```
        @Override
        public void onRemoteUserAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFra

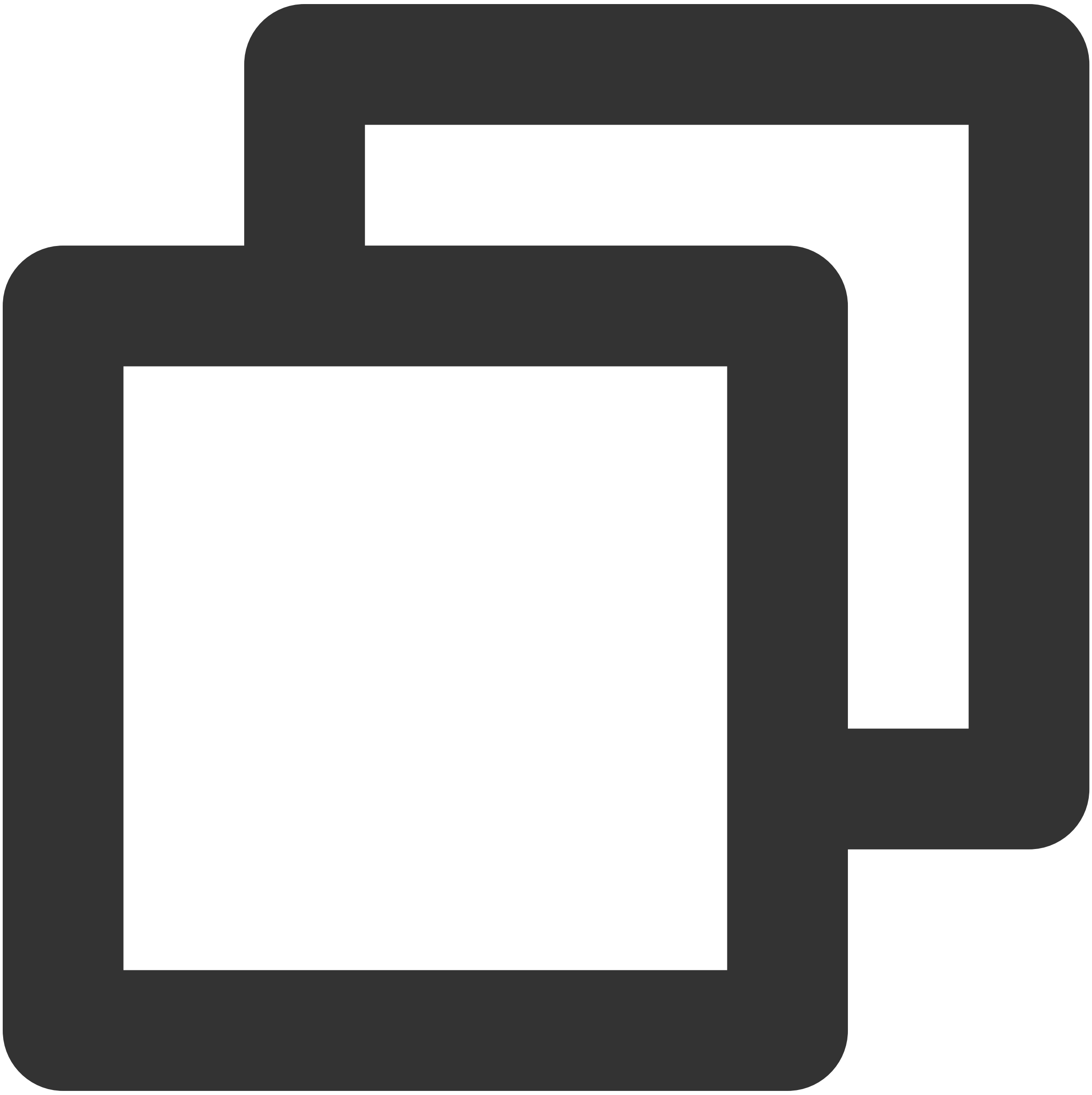
        }

        @Override
        public void onMixedPlayAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFram

        }

        @Override
        public void onMixedAllAudioFrame(TRTCCloudDef.TRTCAudioFrame trtcAudioFrame
            // For more information, see the custom rendering tool class `com.tence
        }
    });
```

}



```
 [self.trtcCloud setAudioFrameDelegate:self];
 // MARK: - TRTCAudioFrameDelegate
 - (void)onCapturedRawAudioFrame:(TRTCAudioFrame *)frame {
        NSLog(@"onCapturedRawAudioFrame");
}

- (void)onLocalProcessedAudioFrame:(TRTCAudioFrame *)frame {
        NSLog(@"onLocalProcessedAudioFrame");
}

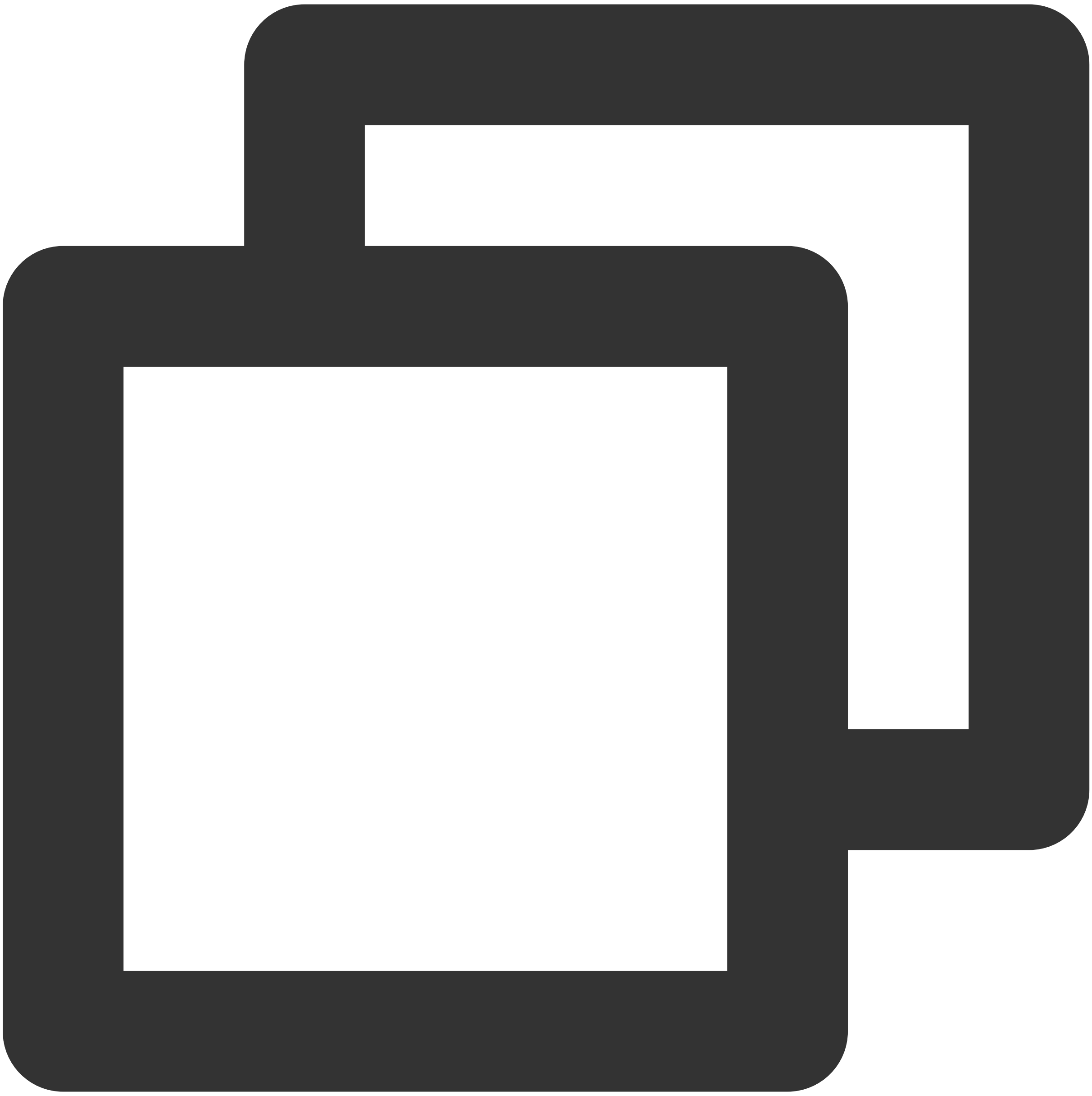
- (void)onRemoteUserAudioFrame:(TRTCAudioFrame *)frame userId:(NSString *)userId {
```



```
        NSLog(@"onRemoteUserAudioFrame");
}

- (void)onMixedPlayAudioFrame:(TRTCAudioFrame *)frame {
        NSLog(@"onMixedPlayAudioFrame");
}

- (void)onMixedAllAudioFrame:(TRTCAudioFrame *)frame {
        NSLog(@"onMixedAllAudioFrame");
}
```



```
// Set custom audio data callback
liteav::ITRTCCloud* trtc_cloud = liteav::ITRTCCloud::getTRTCShareInstance();
trtc_cloud->setAudioFrameCallback(callback)

// Callback APIs for custom audio

virtual void onCapturedRawAudioFrame(TRTCAudioFrame* frame) {
}

virtual void onLocalProcessedAudioFrame(TRTCAudioFrame* frame) {
}

virtual void onPlayAudioFrame(TRTCAudioFrame* frame, const char* userId) {
}

virtual void onMixedPlayAudioFrame(TRTCAudioFrame* frame) {
}
```

notice

Do not perform time-consuming operations with any of the above callback functions. We recommend that you copy the data to another thread for processing to avoid AEC failure and choppy audio.

The data called back by the above callback functions can only be read and copied. Modifying the data may lead to unexpected results.

Web

Last updated：2023-10-08 15:56:46

Function Description

This article mainly introduces the advanced usage of custom capture and custom rendering.

Custom Capture

By default, `trtc.startLocalVideo()` and `trtc.startLocalAudio()` enable camera and microphone capture.

If you need to customize the capture, you can specify the `option.videoTrack` / `option.audioTrack` parameter of the `trtc.startLocalVideo()` / `trtc.startLocalAudio()` method.

There are usually several ways to obtain `audioTrack` and `videoTrack` :

Use getUserMedia to capture the camera and microphone.

Use getDisplayMedia to capture screen sharing.

Use `videoElement.captureStream` to capture the audio and video being played in the video tag.

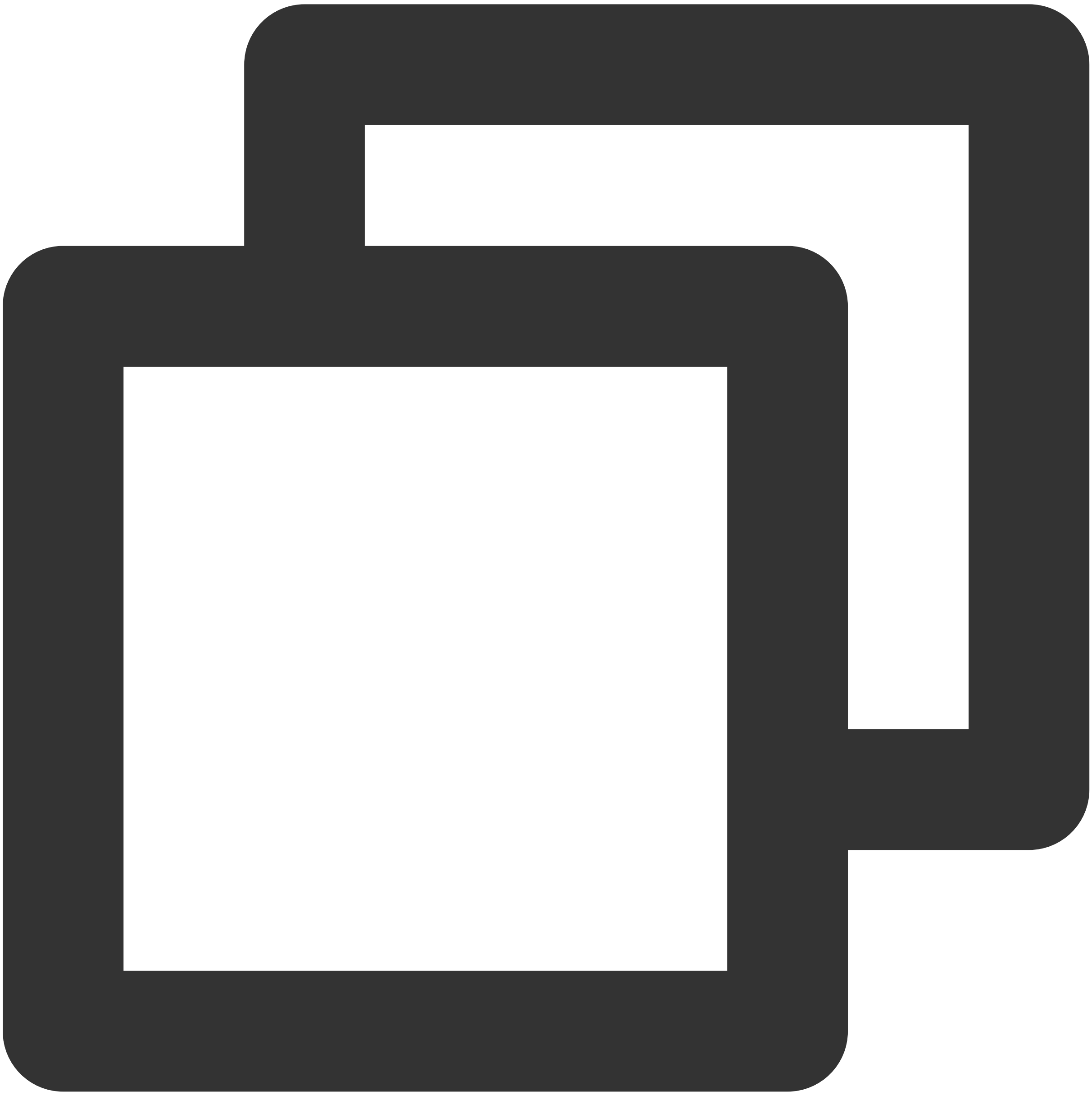
Use `canvas.captureStream` to capture the animation in the canvas.

Capture the video being played in the video tag

```
// Check if your current browser supports capturing streams from video elements
if (!HTMLVideoElement.prototype.captureStream) {
  console.log('your browser does not support capturing stream from video element');
  return
}
// Get the video tag that is playing video on your page
const video = document.getElementByID('your-video-element-ID');
// Capture the video stream from the playing video
const stream = video.captureStream();
const audioTrack = stream.getAudioTracks()[0];
const videoTrack = stream.getVideoTracks()[0];
```

```
trtc.startLocalVideo({ option:{ videoTrack } });
trtc.startLocalAudio({ option:{ audioTrack } });
```

Capture the animation in the canvas



```
// Check if your current browser supports capturing streams from canvas elements
if (!HTMLCanvasElement.prototype.captureStream) {
  console.log('your browser does not support capturing stream from canvas element')
  return
```

```
}
// Get your canvas tag
const canvas = document.getElementByID('your-canvas-element-ID');

// Capture a 15 fps video stream from the canvas
const fps = 15;
const stream = canvas.captureStream(fps);
const videoTrack = stream.getVideoTracks()[0];

trtc.startLocalVideo({ option:{ videoTrack } });
```

Custom Rendering

By default, when calling `trtc.startLocalVideo(view)` or `trtc.startRemoteVideo(view, streamType, userId)` , you need to pass in the `view` parameter. The SDK will create a `video` tag under the specified element tag to play the video.

If you need to customize the rendering, and do not need the SDK to play the video, you can refer to the following steps:

Do not fill in the `view` parameter or pass in `null` when calling the `startLocalVideo` or `startRemoteVideo` method.

Use the `trtc.getVideoTrack(userId, streamType)` method to obtain the corresponding `videoTrack` .

Use your own player for video rendering.

After using this custom rendering method, the `EVENT.VIDEO_PLAY_STATE_CHANGED` event will not be triggered. You need to listen to the `mute/unmute/ended` events of the video track `MediaStreamTrack` to determine the status of the current video data stream.

For remote video, you also need to listen to the `EVENT.REMOTE_VIDEO_AVAILABLE` and `EVENT.REMOTE_VIDEO_UNAVAILABLE` events to handle the lifecycle of remote video.

Custom rendering of local video

```
await trtc.startLocalVideo();
const videoTrack = trtc.getVideoTrack();

// Use your own player for video rendering
const videoElement = document.getElementById('video-element');
videoElement.srcObject = new MediaStream([videoTrack]);
videoElement.play();
```

Custom rendering of remote video

```
trtc.on(TRTC.EVENT.REMOTE_VIDEO_AVAILABLE, async ({ userId, streamType }) => {
  // Only pull the stream, do not play it
  await trtc.startRemoteVideo({ userId, streamType })
  const videoTrack = trtc.getVideoTrack({ userId, streamType });

  // Use your own player for video rendering
  const videoElement = document.getElementById('remote-video-element');
  videoElement.srcObject = new MediaStream([videoTrack]);
  videoElement.play();
});
```

Sending and Receiving Messages

Last updated：2023-10-08 15:57:18

Overview

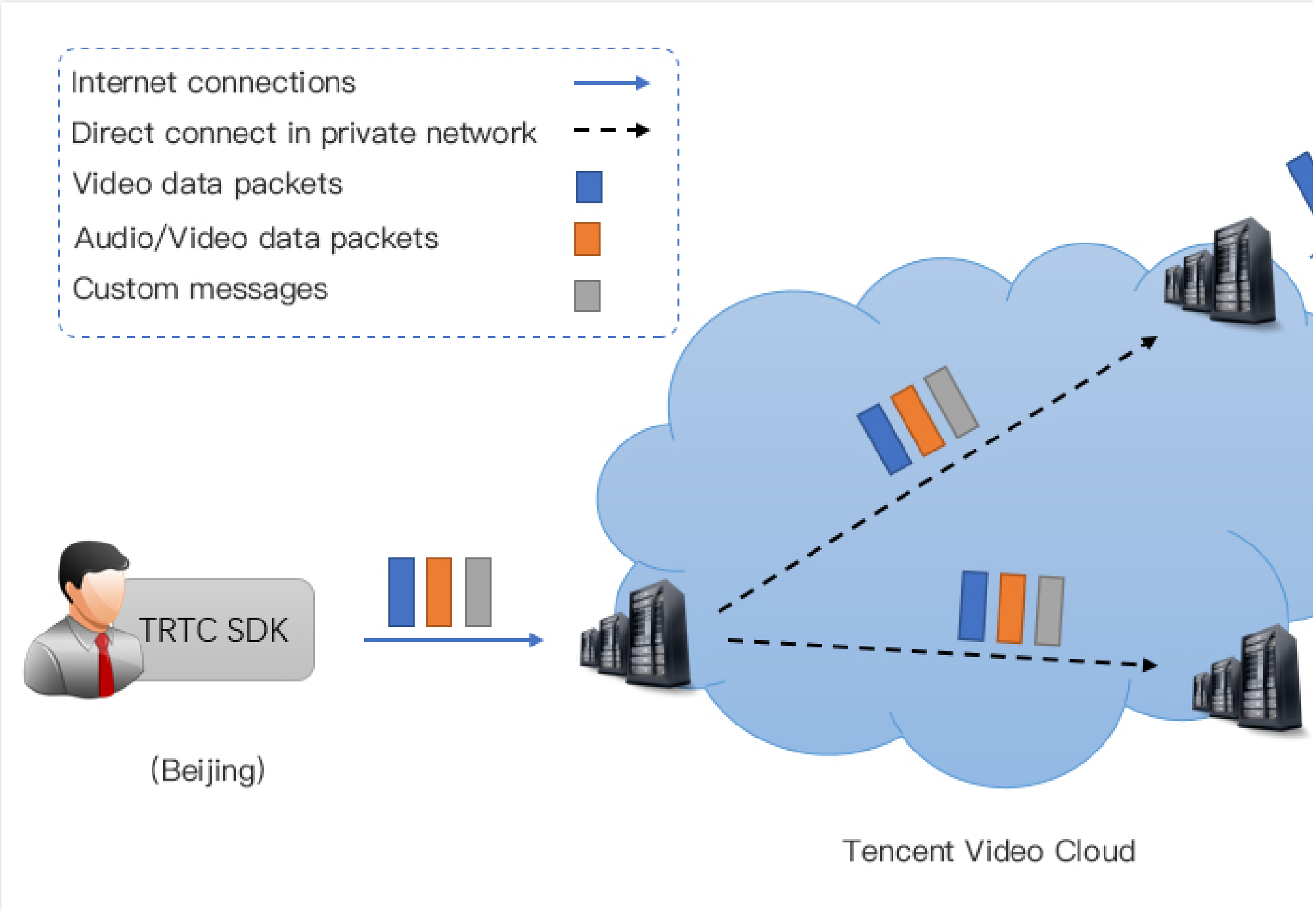
The TRTC SDK provides the ability to send custom messages. With this feature, any user whose role is an anchor can broadcast their own custom messages to other users in the same video room.

Supported Platforms

iOS	Android	macOS	Windows	Electron	web
✓	✓	✓	✓	✓	×

How It Works

A user's custom message will be combined into the audio/video data streams and transmitted to other users in the same room alongside. As audio/video channels themselves are not completely reliable, in order to improve the reliability, the TRTC SDK implements a reliability guarantee mechanism internally.



Sending Messages

Messages are sent by calling the `sendCustomCmdMsg` API of TRTCCloud, and the following four parameters need to be specified during sending:

Parameter Name	Description
cmdID	Message ID. Value range: 1–10. Messages in different business types should use different `cmdIDs` .
data	Message to be sent, which can contain up to 1 KB (1,000 bytes) of data.
reliable	Whether reliable sending is enabled; if yes, the receiver needs to temporarily store the data of a certain period to wait for re-sending, which will cause certain delay.
ordered	Whether orderly sending is enabled, i.e., whether the data should be received in the same order in which it is sent; if yes, the receiver needs to temporarily store and sort messages, which will cause certain delay.

notice

`reliable` and `ordered` must be set to the same value (`YES` or `NO`) and cannot be set to different values currently.

Objective-C

Java

C++

C#

```
// Sample code for sending a custom message
- (void)sendHello {
```

```
    // Command word for the custom message. A set of rules needs to be customized a
    NSInteger cmdID = 0x1;
    NSData *data = [@"Hello" dataUsingEncoding:NSUTF8StringEncoding];
    // `reliable` and `ordered` need to be consistent for now. Orderly sending is u
    [trtcCloud sendCustomCmdMsg:cmdID data:data reliable:YES ordered:YES];
}
```

```
// Sample code for sending a custom message
public void sendHello() {
```

```
        try {
            // Command word for the custom message. A set of rules needs to be customiz
            int cmdID = 0x1;
            String hello = "Hello";
            byte[] data = hello.getBytes("UTF-8");
            // `reliable` and `ordered` need to be consistent for now. Orderly sending
            trtcCloud.sendCustomCmdMsg(cmdID, data, true, true);

        } catch (UnsupportedEncodingException e) {
            e.printStackTrace();
        }
    }
```

```
// Sample code for sending a custom message
void sendHello()
{
    // Command word for the custom message. A set of rules needs to be customized a

    uint32_t cmdID = 0x1;
    uint8_t* data = { '1', '2', '3' };
    uint32_t dataSize = 3;  // Length of data

    // `reliable` and `ordered` need to be consistent for now. Orderly sending is u
    trtcCloud->sendCustomCmdMsg(cmdID, data, dataSize, true, true);
```

```
}
```

```
// Sample code for sending a custom message
private void sendHello()
{
    // Command word for the custom message. A set of rules needs to be customized a

    uint cmdID = 0x1;
    byte[] data = { '1', '2', '3' };
```

```
    uint dataSize = 3;  // Length of data

    // `reliable` and `ordered` need to be consistent for now. Orderly sending is u
    mTRTCCloud.sendCustomCmdMsg(cmdID, data, dataSize, true, true);
}
```

Receiving Messages

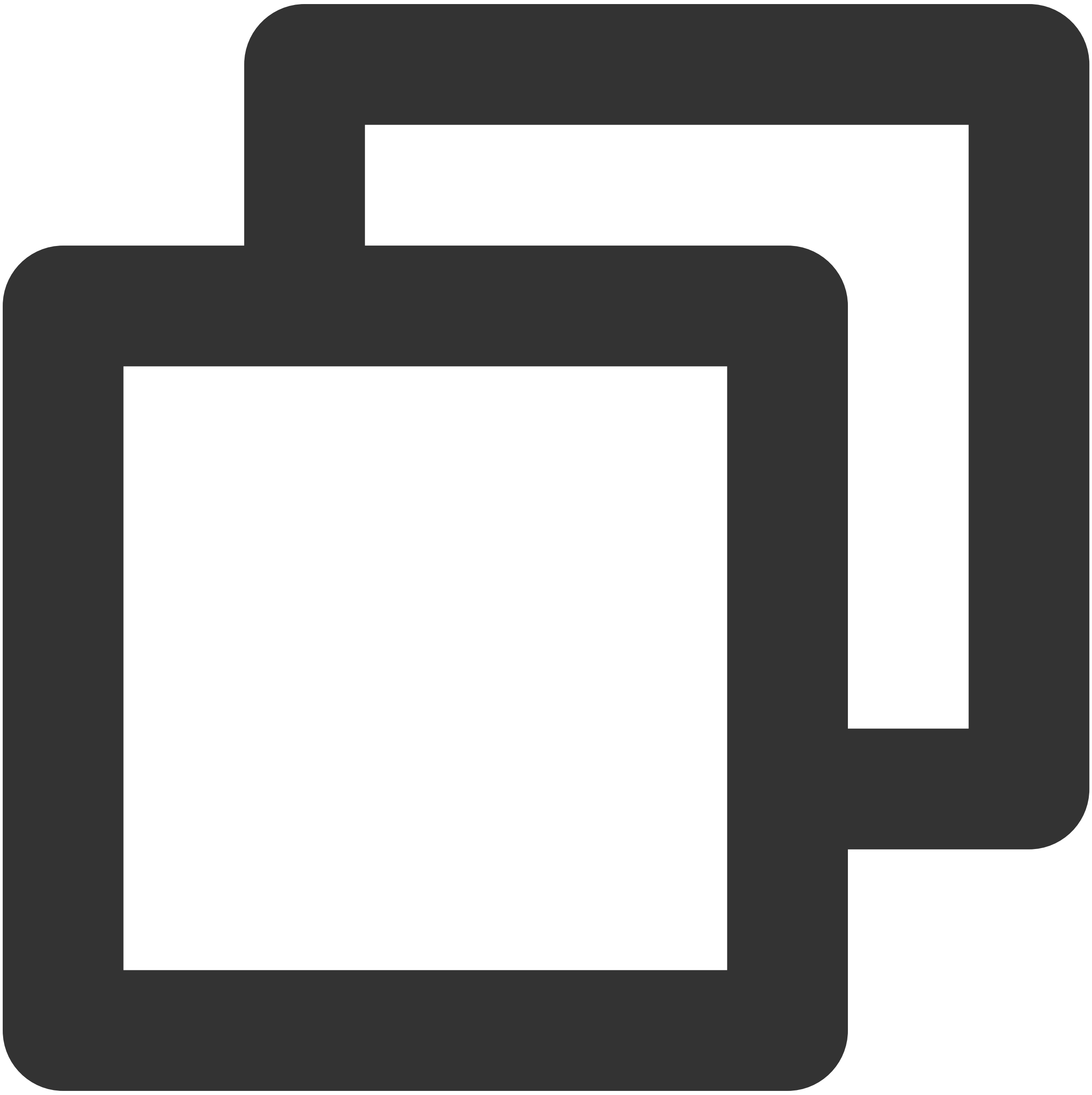
After a user in a room uses `sendCustomCmdMsg` to send a custom message, other users in the room can receive the message through the `onRecvCustomCmdMsg` API in the SDK callback.

Objective-C

Java

C++

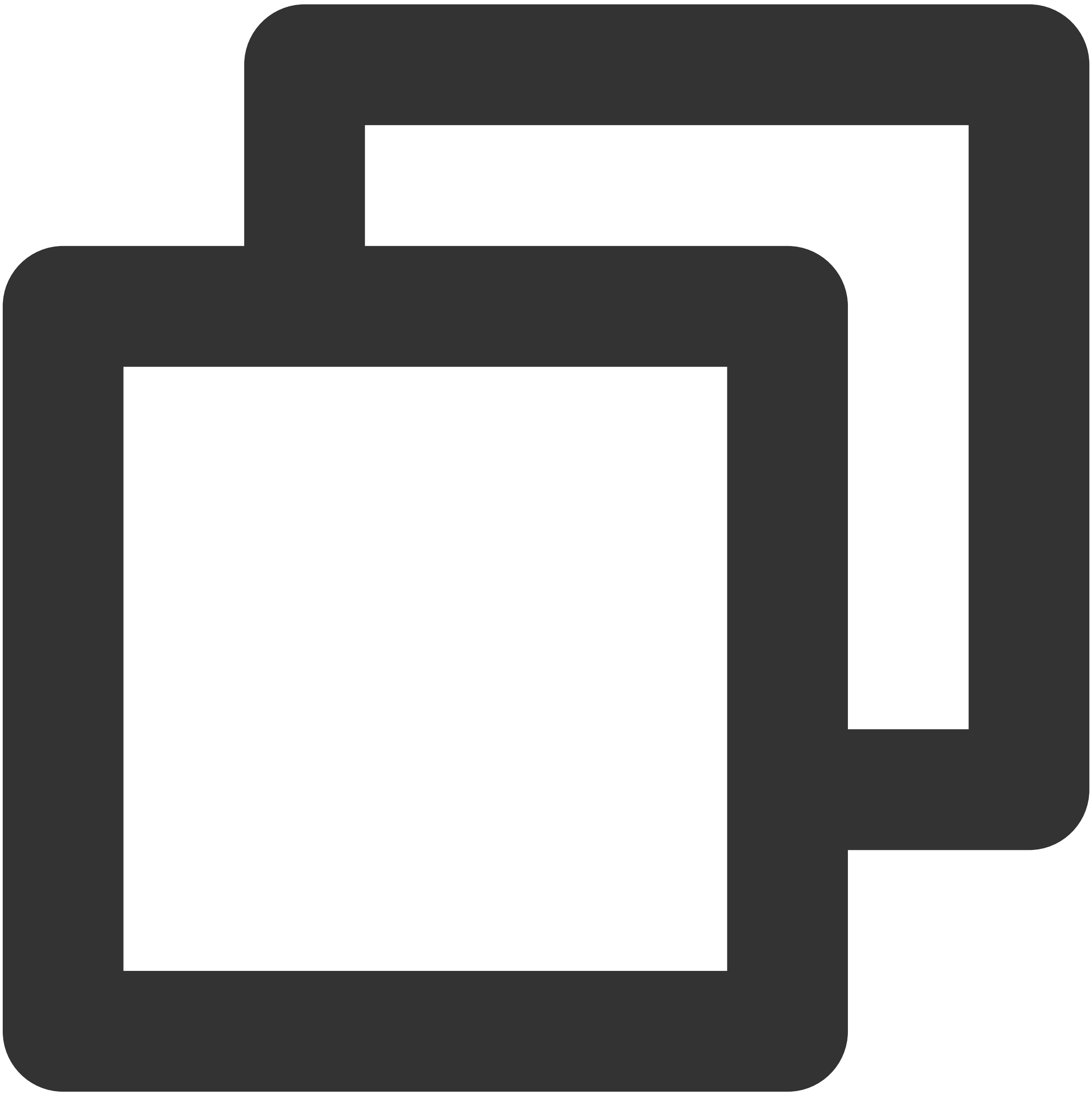
C#



```
// Receive and process messages sent by other users in the room
- (void)onRecvCustomCmdMsgUserId:(NSString *)userId cmdID:(NSInteger)cmdId seq:(UIn
{
    // Receive the message sent by `userId`
    switch (cmdId)  // `cmdId` agreed upon between sender and receiver
    {
    case 0:
        // Process the message with `cmdId` = 0
        break;
    case 1:
        // Process the message with `cmdId` = 1
```



```
            break;
        case 2:
            // Process the message with `cmdId` = 2
            break;
        default:
            break;
        }
    }
```



```
// Inherit `TRTCCloudListener` and implement the `onRecvCustomCmdMsg` method to rec
public void onRecvCustomCmdMsg(String userId, int cmdId, int seq, byte[] message) {
    // Receive the message sent by `userId`
    switch (cmdId)  // `cmdId` agreed upon between sender and receiver
    {
    case 0:
        // Process the message with `cmdId` = 0
        break;
    case 1:
        // Process the message with `cmdId` = 1
        break;
    case 2:
        // Process the message with `cmdId` = 2
        break;
    default:
        break;

}
```

```
// Receive and process messages sent by other users in the room
void TRTCCloudCallbackImpl::onRecvCustomCmdMsg(
                            const char* userId, int32_t cmdId, uint32_t seq, const
{
    // Receive the message sent by `userId`
    switch (cmdId)  // `cmdId` agreed upon between sender and receiver
    {
    case 0:
        // Process the message with `cmdId` = 0
        break;
    case 1:
```



```
            // Process the message with `cmdId` = 1
            break;
        case 2:
            // Process the message with `cmdId` = 2
            break;
        default:
            break;
        }
}
```

```
// Receive and process messages sent by other users in the room
public void onRecvCustomCmdMsg(string userId, int cmdId, uint seq, byte[] msg, uint
{
    // Receive the message sent by `userId`
    switch (cmdId)  // `cmdId` agreed upon between sender and receiver
    {
    case 0:
        // Process the message with `cmdId` = 0
        break;
    case 1:
        // Process the message with `cmdId` = 1
        break;
    case 2:
        // Process the message with `cmdId` = 2
        break;
    default:
        break;
    }
}
```

Use Limits

Since custom messages have a higher transmission priority than audio/video data, if too many of them are sent, audio/video data may be interfered with, resulting in video lagging or blurring. Therefore, the following frequency limits apply to custom messages:

As custom messages are broadcast to all users in the same room, up to 30 messages can be sent per second.

A data packet (i.e., data size) can be of up to 1 KB; if the threshold is exceeded, the packet is very likely to be discarded by the intermediate router or server.

A client can send up to 8 KB of data in total per second, that is, if each data packet is of 1 KB, up to 8 packets can be sent per second.

Event Callbacks

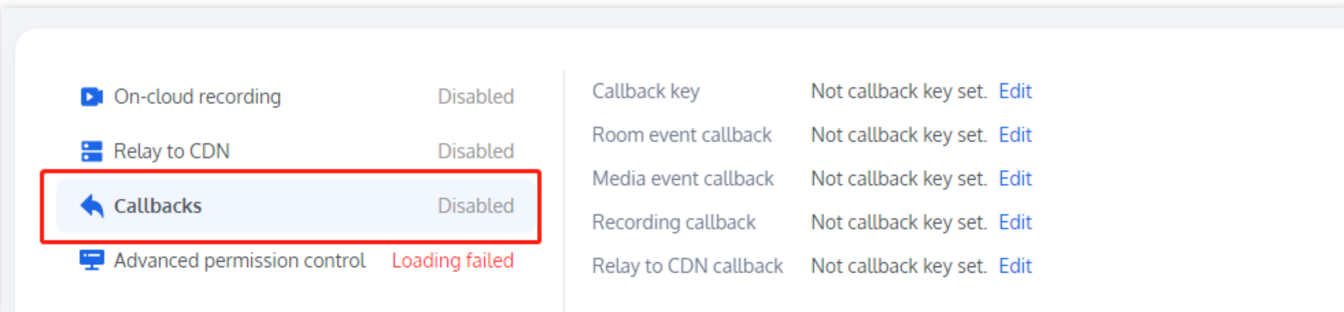
Event Callbacks

Last updated：2024-08-07 10:53:53

The event callback service can send notifications about TRTC events in the form of HTTP/HTTPS requests to your server. Currently, you can register callbacks for room events, media events, as well as some recording events (for details about on-cloud recording callbacks, see On-Cloud Recording). To receive such callbacks, you need to configure callback information in the TRTC console.

Callback Information

In order to receive event callback notifications, you need to configure callback information in Tencent RTC Console.



Note:

You need to provide the following information:

Required: An HTTP/HTTPS server address to receive callback notifications.

Optional: A custom key containing up to 32 uppercase and lowercase letters and digits, which is needed for the calculation of signatures.

Timeout and Retry

A notification will be considered failed if the callback server does not receive a response from your server within five seconds of message sending. It will try again immediately after the first failure and retry **10 seconds** after every subsequent failure. The retries will stop one minute after the first try.

Format of Callback Messages

Callbacks are sent to your server in the form of HTTP/HTTPS POST requests.

Character encoding: UTF-8

Request: JSON for the request body

Response: HTTP STATUS CODE = 200. The server ignores the content of the response packet. For protocol-friendliness, we recommend adding `JSON:` {"code":0}` to the response.

Parameters

Callback parameters

The header of a callback message contains the following fields.

Field	Value
Content-Type	application/json
Sign	The signature value.
SdkAppId	The SDK application ID.

The body of a callback message contains the following fields.

Field	Type	Description
EventGroupId	Number	The event group ID.
EventType	Number	The type of the callback event.
CallbackTs	Number	The Unix timestamp (ms) of callback sending.
EventInfo	JSON Object	The event information.

Event group ID

Field	Value	Description
EVENT_GROUP_ROOM	1	Room event group
EVENT_GROUP_MEDIA	2	Media event group

Note:

For on-cloud recording events, see On-Cloud Recording.

Event type

Field	Value	Description
EVENT_TYPE_CREATE_ROOM	101	Creating room
EVENT_TYPE_DISMISS_ROOM	102	Closing room
EVENT_TYPE_ENTER_ROOM	103	Entering room
EVENT_TYPE_EXIT_ROOM	104	Leaving room
EVENT_TYPE_CHANGE_ROLE	105	Switching roles
EVENT_TYPE_START_VIDEO	201	Starting pushing video data
EVENT_TYPE_STOP_VIDEO	202	Stopping pushing video data
EVENT_TYPE_START_AUDIO	203	Starting pushing audio data
EVENT_TYPE_STOP_AUDIO	204	Stopping pushing audio data
EVENT_TYPE_START_ASSIT	205	Starting pushing substream data
EVENT_TYPE_STOP_ASSIT	206	Stopping pushing substream data

Note:

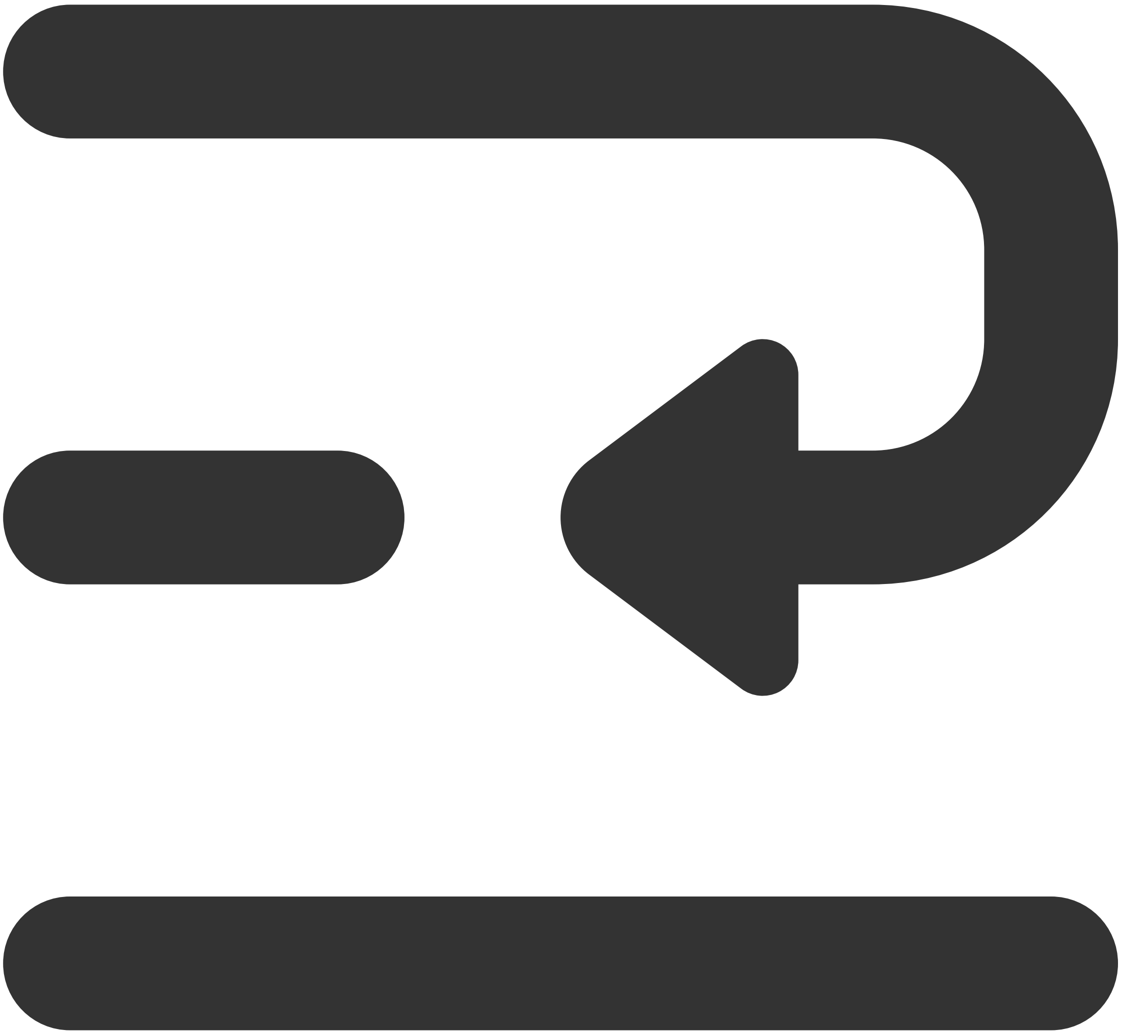
Room exit will trigger only the `104` callback and not the `202` or `204` callback. `202` and `204` are triggered only if a user manually turns their video and audio off.

Event Callback Example

101
102
103
104
105
201
202
203
204
205
206

```
        }
}
```


```
{       "EventGroupId": 1,
        "EventType":    101,
        "CallbackTs":   1687770730166,
        "EventInfo":    {
                "RoomId":       12345,
        "EventTs":      1687770730,
        "EventMsTs":    1687770730160,
        "UserId":       "test"
        }
}
```

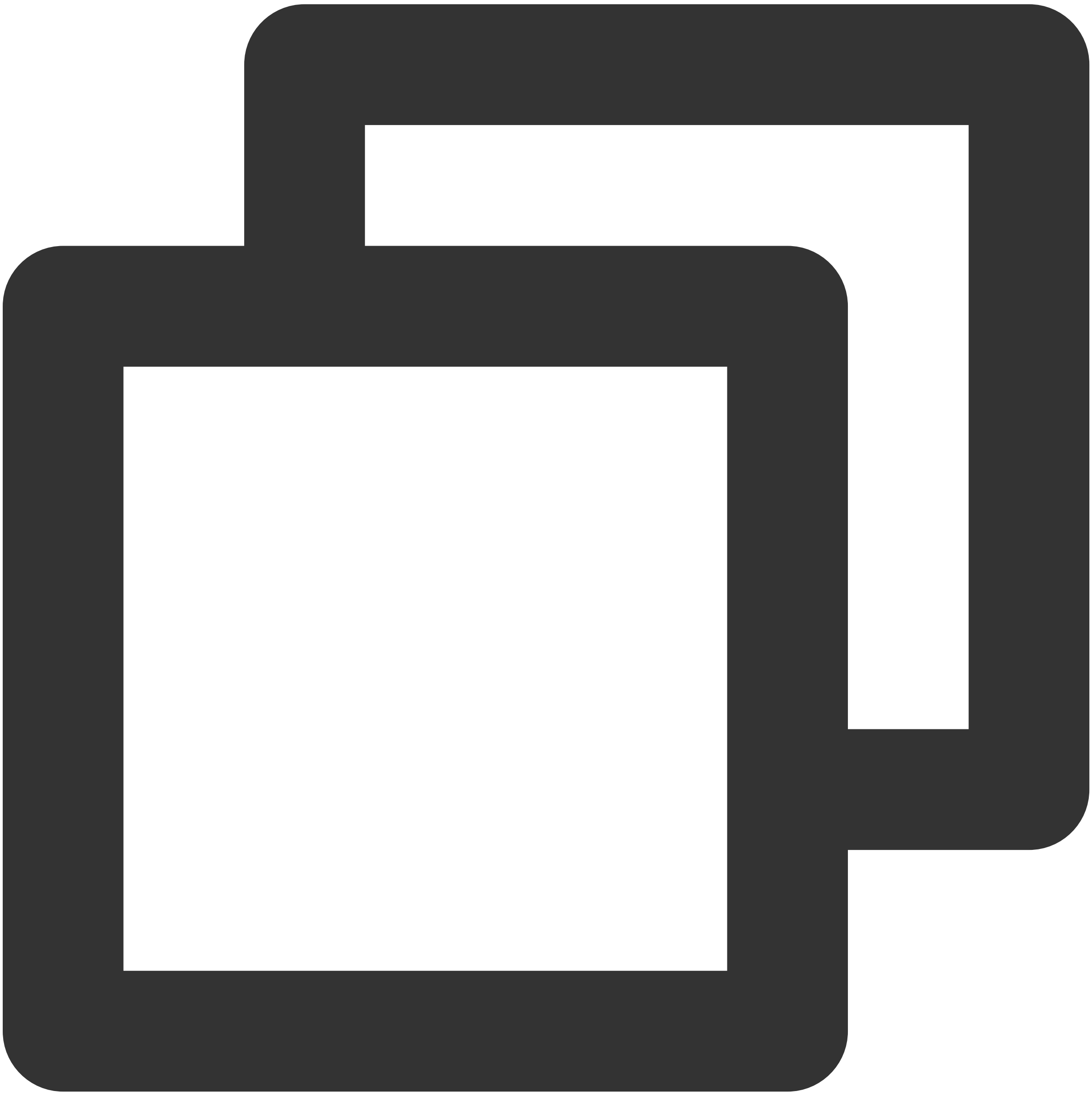


}


```
{
    "EventGroupId":     1,
    "EventType":        102,
    "CallbackTs":       1687771618531,
    "EventInfo":        {
        "RoomId":       "12345",
        "EventTs":      1687771618,
        "EventMsTs":    1687771618457
     }
}
```

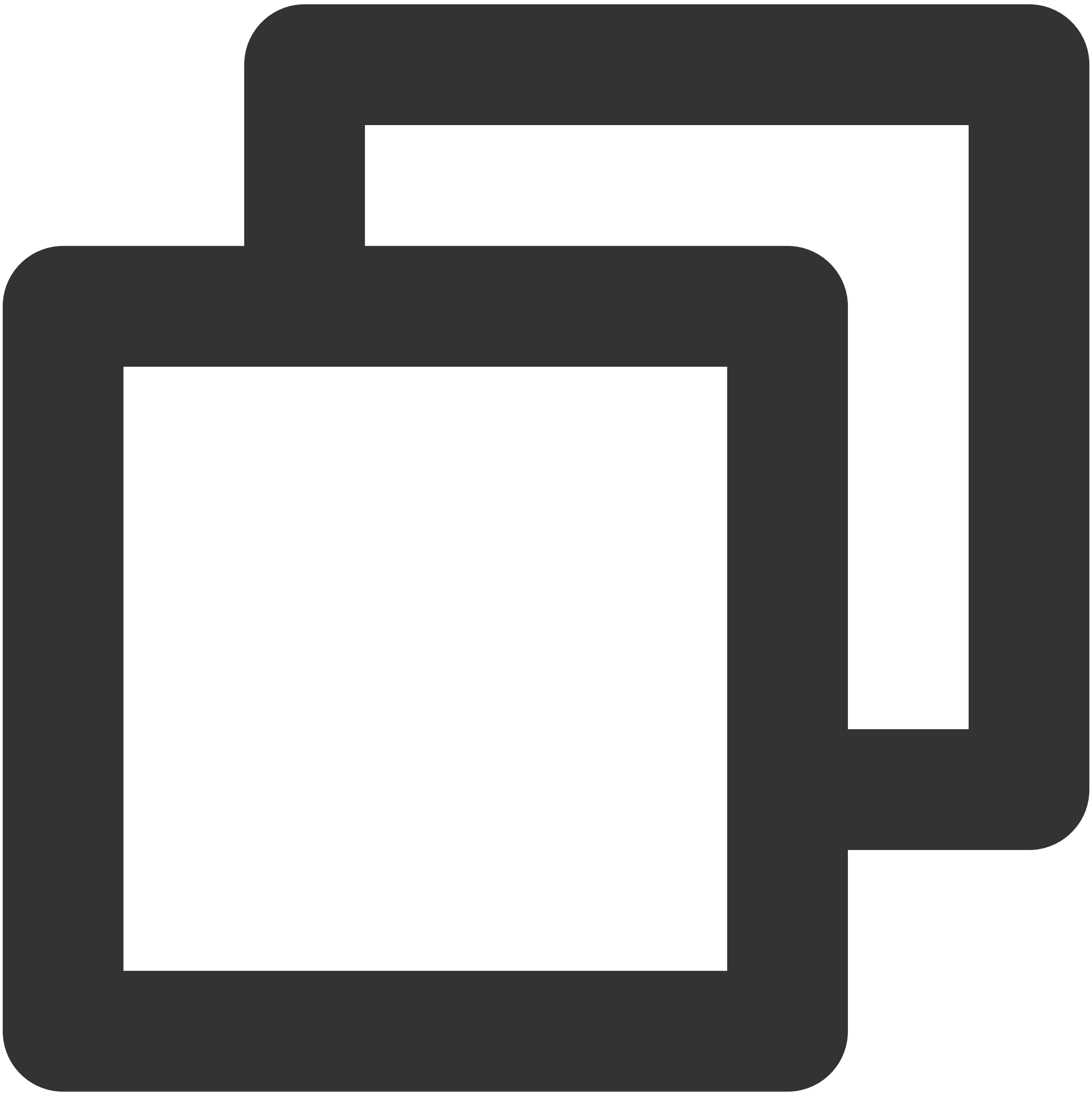
```
{
        "EventGroupId": 1,
        "EventType":    103,
        "CallbackTs":   1687770731932,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687770731,
                "EventMsTs":    1687770731831,
                "UserId":       "test",
                "Role": 21,
                "TerminalType": 2,
```

```
                "UserType":     3,
                "Reason":       1
        }
}
```



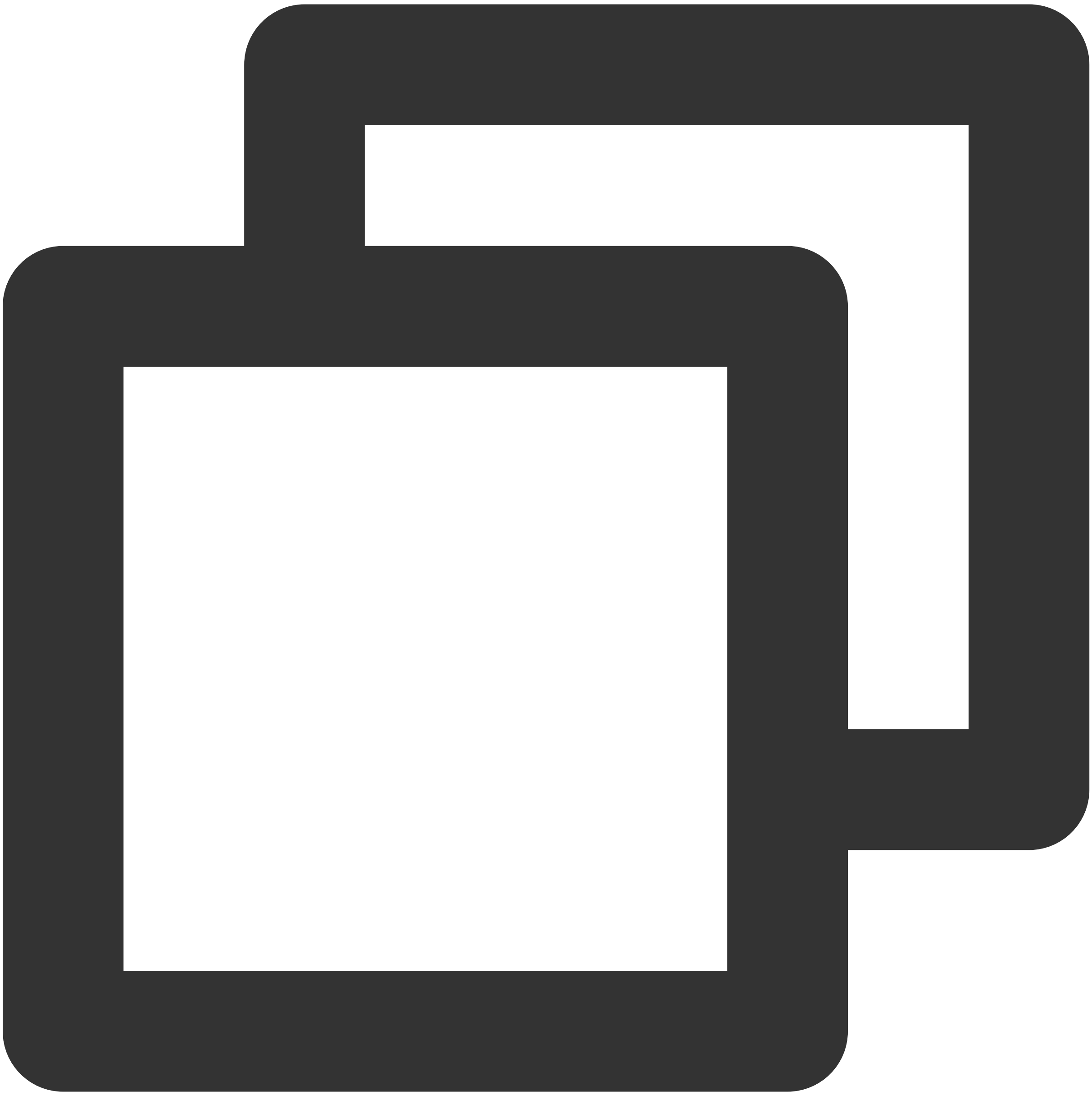
```
{
        "EventGroupId": 1,
        "EventType":    104,
        "CallbackTs":   1687770731922,
        "EventInfo":    {
                "RoomId":       12345,
```

```
                "EventTs":      1687770731,
                "EventMsTs":    1687770731898,
                "UserId":       "test",
                "Role": 20,
                "Reason":       1
        }
}
```



```
{
        "EventGroupId": 1,
        "EventType":    105,
```

```
        "CallbackTs":   1687772245596,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687772245,
                "EventMsTs":    1687772245537,
                "UserId":       "test",
                "Role": 21
        }
}
```



```
{
```



```
        "EventGroupId": 2,
        "EventType":    201,
        "CallbackTs":   1687771803198,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687771803,
                "EventMsTs":    1687771803192,
                "UserId":       "test"
        }
}
```



```
{
        "EventGroupId": 2,
        "EventType":    202,
        "CallbackTs":   1687771919458,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687771919,
                "EventMsTs":    1687771919447,
                "UserId":       "test",
                "Reason":       0
        }
}
```

}

```
{
        "EventGroupId": 2,
        "EventType":    203,
        "CallbackTs":   1687771869377,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687771869,
                "EventMsTs":    1687771869365,
                "UserId":       "test"
        }
}
```

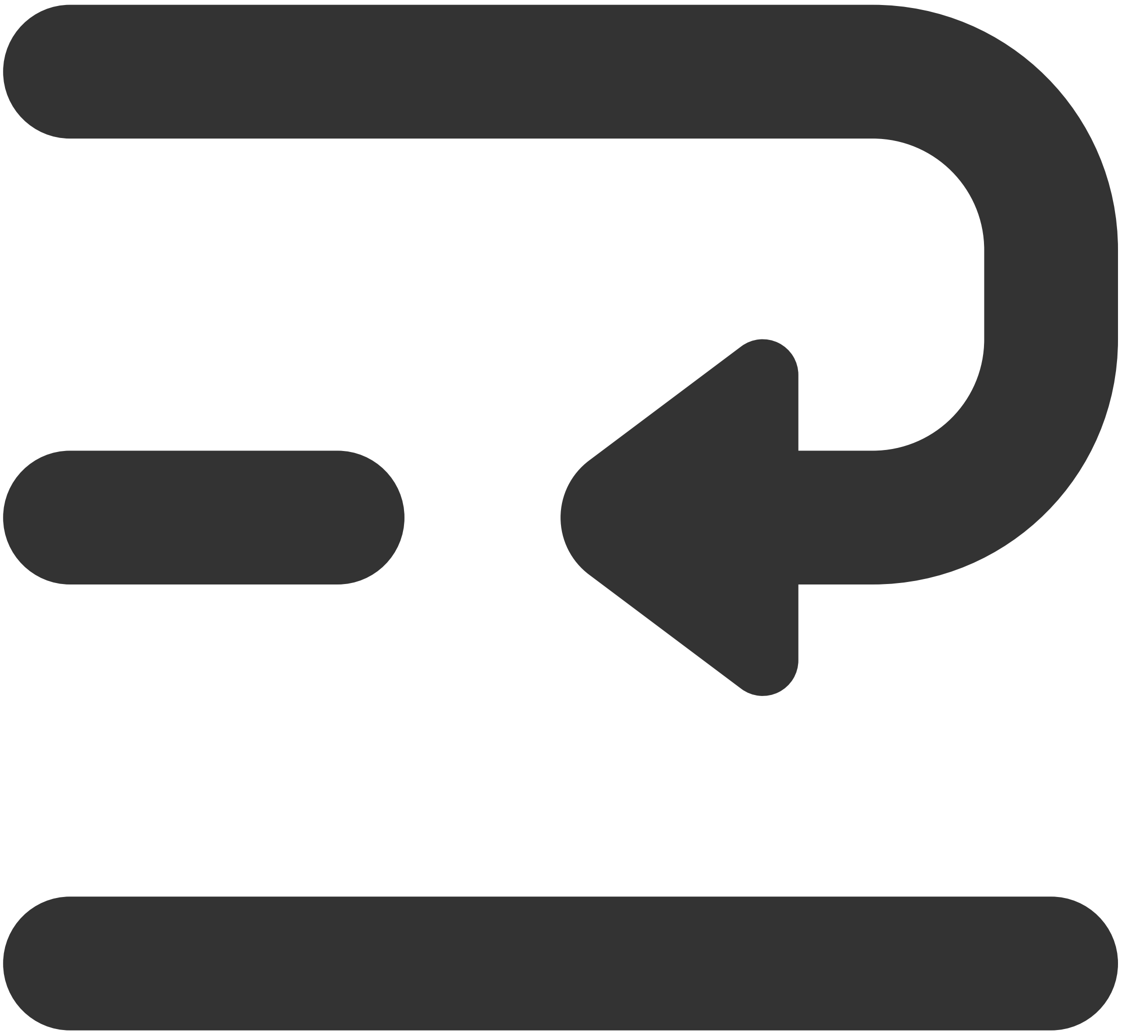
```
{
        "EventGroupId": 2,
        "EventType":    204,
        "CallbackTs":   1687770732498,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687770732,
                "EventMsTs":    1687770732383,
                "UserId":       "test",
                "Reason":       0
```



```
        }
}
```

```
{
        "EventGroupId": 2,
        "EventType":    205,
        "CallbackTs":   1687772013823,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687772013,
                "EventMsTs":    1687772013753,
```

```
                    "UserId":        "test"
          }
}
```



```
{
        "EventGroupId": 2,
        "EventType":    206,
        "CallbackTs":   1687772015054,
        "EventInfo":    {
                "RoomId":       12345,
                "EventTs":      1687772015,
                "EventMsTs":    1687772015032,
                "UserId":       "test",
                "Reason":       0
        }
```

```
}
```

Event information

Field	Type	Description
RoomId	String/Number	The room ID, which is of the same type as the room ID on the client.
EventTs	Number	The Unix timestamp (seconds) of event occurrence. This field is reserved for compatibility purposes.
EventMsTs	Number	The Unix timestamp (ms) of event occurrence.
UserId	String	User ID
UniqueId	Number	The unique identifier of an event (optional), which is valid for the room event group. When a user experiences unusual events such as network change or abnormal exit and reentry, your server may receive multiple callbacks for the entry and exit of the same user. A unique identifier helps identify a room entry or exit.
Role	Number	The role type (optional), which is valid for the room entry/exit callback.
TerminalType	Number	The device type (optional), which is valid for the room entry callback.
UserType	Number	The user type (optional), which is valid for the room entry callback.
Reason	Number	The reason (optional), which is valid for the room entry/exit callback.

Note:

We have developed a policy that prevents repeated callbacks resulting from unusual events on the client. If you start using the callback service after July 30, 2021, the policy will apply by default, and the room event group will no longer carry UniqueId.

Role type

Field	Value	Description
MEMBER_TRTC_ANCHOR	20	Anchor
MEMBER_TRTC_VIEWER	21	Audience

Device type

Field	Value	Description

TERMINAL_TYPE_WINDOWS	1	Windows
TERMINAL_TYPE_ANDROID	2	Android
TERMINAL_TYPE_IOS	3	iOS
TERMINAL_TYPE_LINUX	4	Linux
TERMINAL_TYPE_OTHER	100	Other

User type

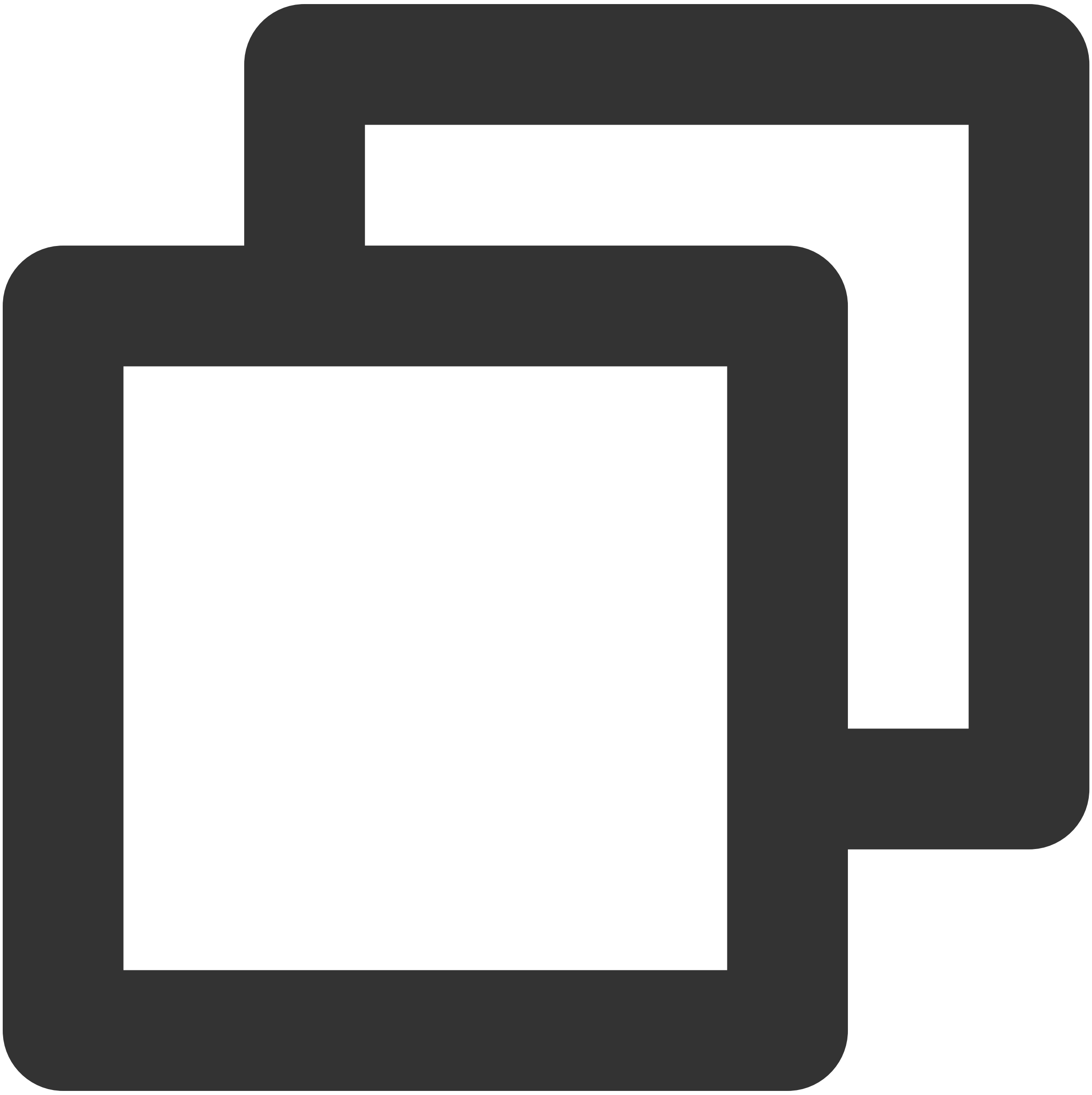
Field	Value	Description
USER_TYPE_WEBRTC	1	WebRTC
USER_TYPE_APPLET	2	Mini Program
USER_TYPE_NATIVE_SDK	3	Native SDK

Reason

Field	Description
Room entry	1: Voluntary entry 2: Network change 3: Timeout and retry 4: Cross-room communication
Room exit	1: Voluntary exit 2: Timeout 3: Removed from the room 4: Cross-room communication was canceled 5: The process was force-closed **Note: TRTC cannot capture a force-close event on Android and will send a callback only after timeout (`reason` = `2`).**

Signature calculation

Signatures are calculated using the HMAC SHA256 encryption algorithm. Upon receiving a callback message, your server will calculate a signature using the same method, and if the results match, it indicates that the callback is from TRTC and not forged. See below for the calculation method.



```
// In the formula below, `key` is the key used to calculate a signature.
Sign = base64(hmacsha256(key, body))
```

Note:

`body` is the original packet body of the callback request you receive. Do not make any modifications. Below is an example.

```
body="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t103,\\n\\t\\"Callback
```

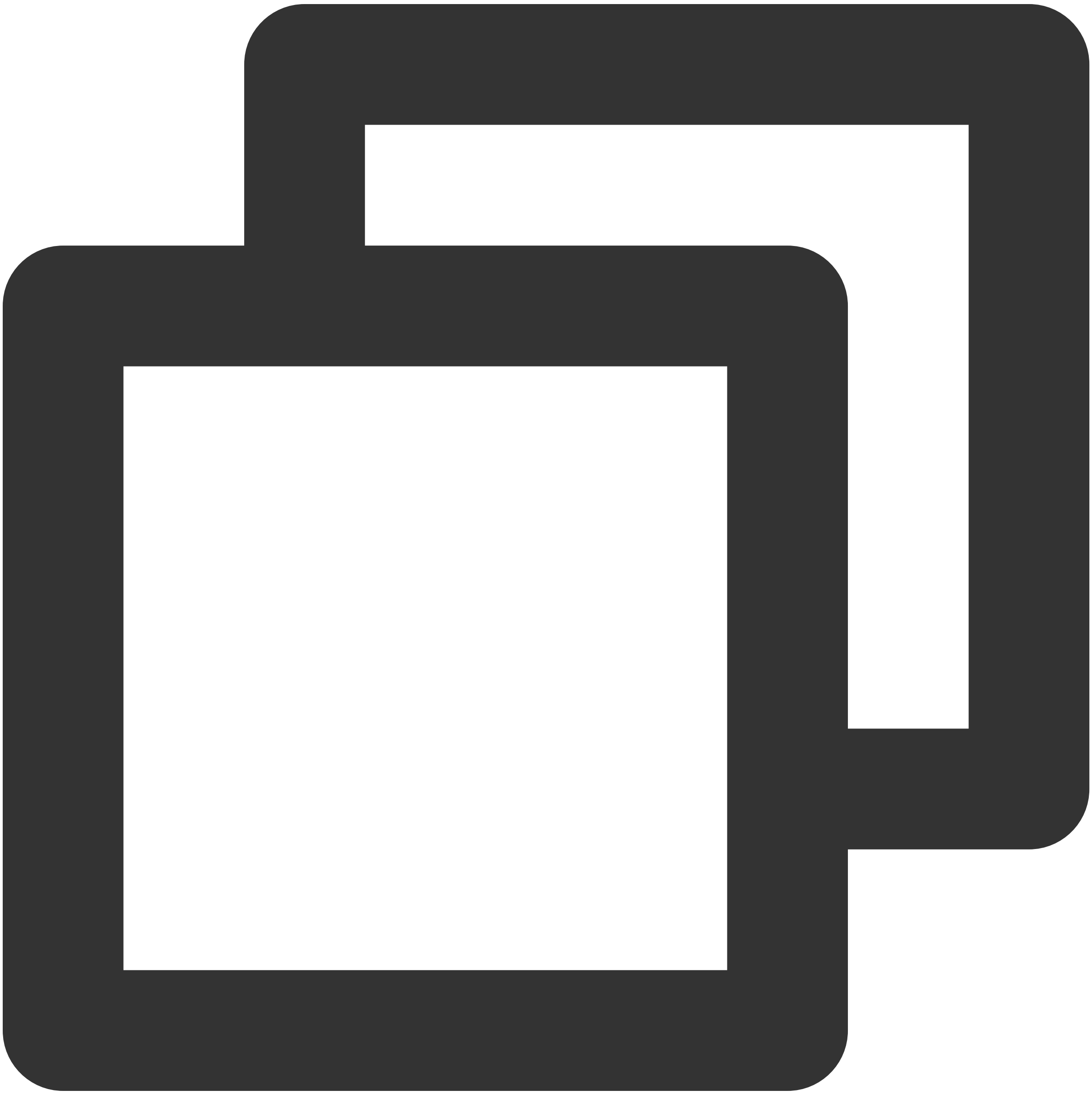
Verify signature example

Java

Python

PHP

Golang



```
import javax.crypto.Mac;
import javax.crypto.spec.SecretKeySpec;
import java.util.Base64;
//# Function: Third-party callback sign verification
//# Parameters:
//#   key: The key configured in the console
//#   body: The body returned by the Tencent Cloud callback
//#   sign: The sign value returned by the Tencent Cloud callback
//# Return Value:
//#   Status: OK indicates that the verification has passed, FAIL indicates that th
//#   Info: Success/Failure information
```

```
public class checkSign {
    public static String getResultSign(String key, String body) throws Exception {
        Mac hmacSha256 = Mac.getInstance("HmacSHA256");
        SecretKeySpec secret_key = new SecretKeySpec(key.getBytes(), "HmacSHA256");
        hmacSha256.init(secret_key);
        return Base64.getEncoder().encodeToString(hmacSha256.doFinal(body.getBytes(
    }
    public static void main(String[] args) throws Exception {
        String key = "123654";
        String body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\
        String Sign = "kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA=";
        String resultSign = getResultSign(key, body);

        if (resultSign.equals(Sign)) {
            System.out.println("{'Status': 'OK', 'Info': 'validation passed'}");
        } else {
            System.out.println("{'Status': 'FAIL', 'Info': 'validation failed'}");
        }
    }
}
```

```
# -*- coding: utf8 -*-
import hmac
import base64
from hashlib import sha256

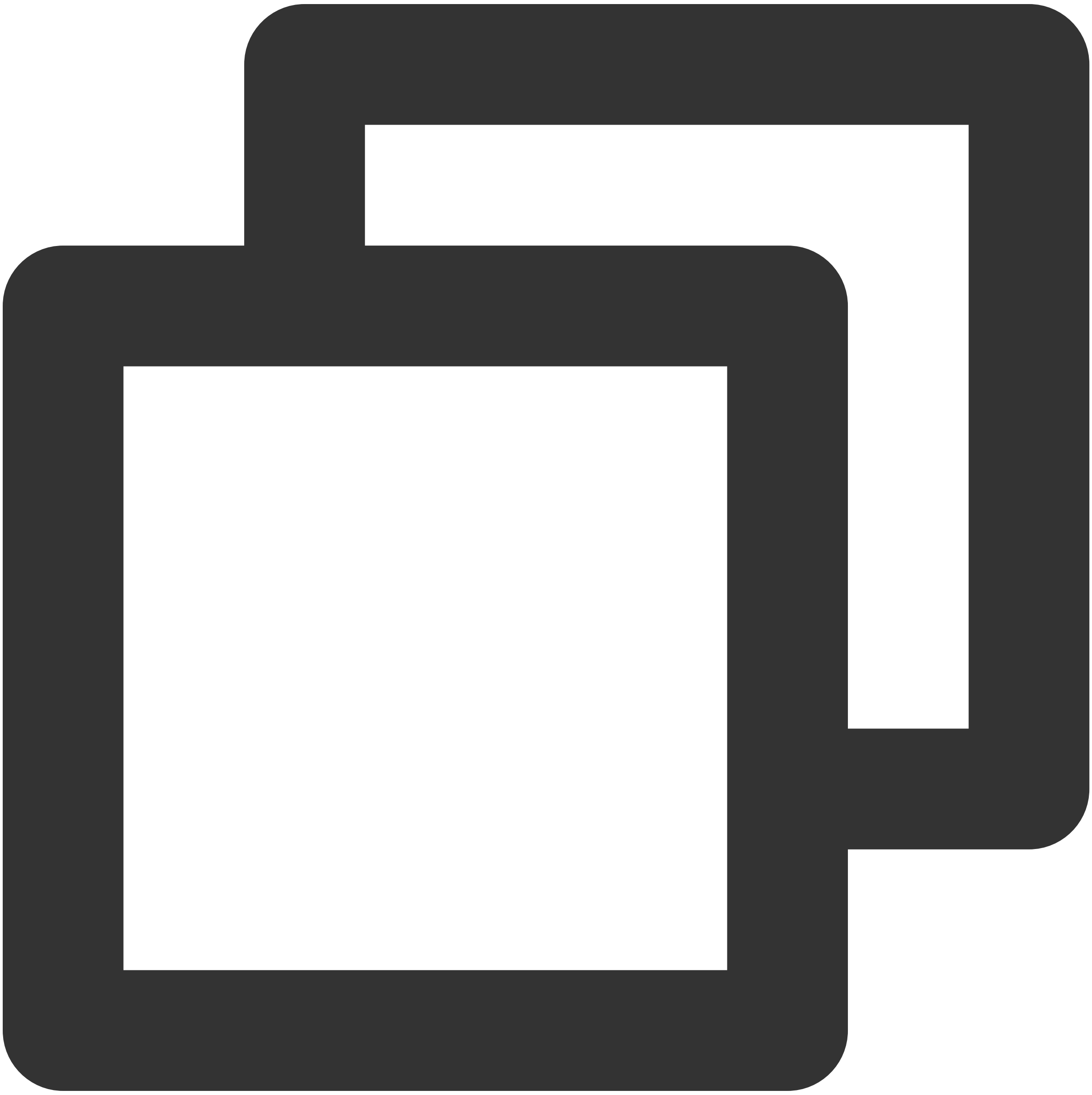
# Function: Third-party callback sign verification
# Parameters:
#   key: The key configured in the console
#   body: The body returned by the Tencent Cloud callback
#   sign: The sign value returned by the Tencent Cloud callback
# Return Value:
```



```
#   Status: OK indicates that the verification has passed, FAIL indicates that the
#   Info: Success/Failure information

def checkSign(key, body, sign):
    temp_dict = {}
    computSign = base64.b64encode(hmac.new(key.encode('utf-8'), body.encode('utf-8'
    print(computSign)
    if computSign == sign:
        temp_dict['Status'] = 'OK'
        temp_dict['Info'] = 'validation passed'
        return temp_dict
    else:
        temp_dict['Status'] = 'FAIL'
        temp_dict['Info'] = 'validation failed'
        return temp_dict

if __name__ == '__main__':
    key = '123654'
    body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\\":\\t204,\
    sign = 'kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA='
    result = checkSign(key, body, sign)
    print(result)
```



```
<?php

class TlsEventSig {

    private $key = false;
    private $body = false;

    public function __construct( $key, $body ) {
        $this->key = $key;
                $this->body = $body;
    }
```



```
    private function __hmacsha256() {
        $hash = hash_hmac( 'sha256', $this->body, $this->key, true );
                return base64_encode( $hash);
    }

    public function genEventSig() {
        return $this->__hmacsha256();
    }
}

$key="789";
$data="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\\"Callbac

$api = new  TlsEventSig($key, $data);
echo $api->genEventSig();
```

```
package main
import "fmt"
import (
        "crypto/hmac"
        "crypto/sha256"
        "encoding/base64"
)

func main () {
    var data = "{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\
    var key = "789"
```

```
    //JSRUN engine 2.0, supporting up to 30 types of languages for online running,
    fmt.Println(hmacsha256(data,key))
}

func hmacsha256(data string, key string) string {
        h := hmac.New(sha256.New, []byte(key))
        h.Write([]byte(data))
        return base64.StdEncoding.EncodeToString(h.Sum(nil))
}
```

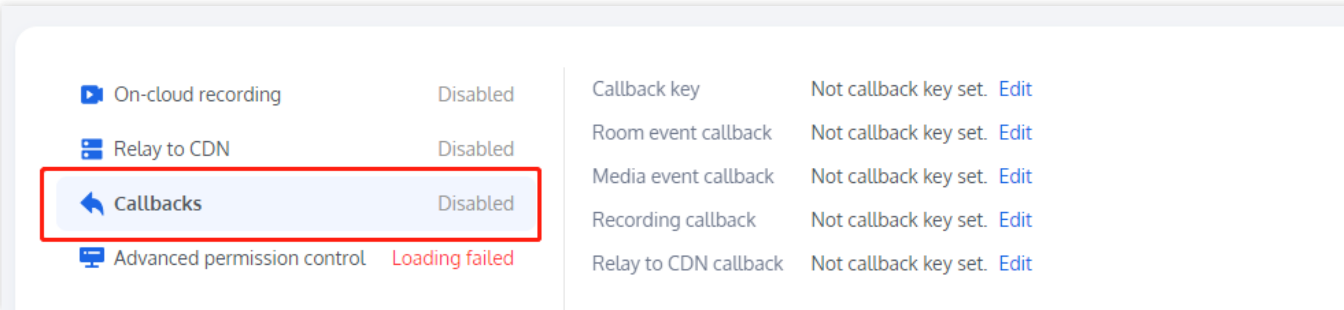
Relay to CDN Callback

Last updated：2024-08-07 10:53:53

The server relay to CDN callback supports notifying your server of the events generated by the relay to CDN REST API in the form of HTTP/HTTPS requests. To receive such callbacks, you need to configure callback information in the TRTC console.

Callback Information

In order to receive event callback notifications, you need to configure callback information in the Tencent RTC Console.



Note：

You need to provide the following information:

Required: An HTTP/HTTPS server address to receive callback notifications.

Optional: A custom key containing up to 32 uppercase and lowercase letters and digits, which is needed for the calculation of signatures.

Timeout and Retry

A notification will be considered failed if the callback server does not receive a response from your server within five seconds of message sending. It will try again immediately after the first failure and retry **10 seconds** after every subsequent failure. The retries will stop one minute after the first try.

Format of Callback Messages

Callbacks are sent to your server in the form of HTTP/HTTPS POST requests.

Character encoding: UTF-8

Request: JSON for the request body

Response: HTTP STATUS CODE = 200. The server ignores the content of the response packet. For protocol-friendliness, we recommend adding `JSON:` {"code":0}` to the response.

Parameters

Callback parameters

The header of a callback message contains the following fields.

Field	Value
Content-Type	application/json
Sign	The signature value.
SdkAppId	The SDK application ID.

The body of a callback message contains the following fields.

Field	Type	Description
EventGroupId	Number	The event group ID, mix relay event fixed as 4.
EventType	Number	The type of the callback event.
CallbackTs	Number	The Unix timestamp (ms) of callback sending.
EventInfo	JSON Object	The event information.

Event group ID

Field	Value	Description
EVENT_GROUP_CLOUD_PUBLISH	4	Relay event group

Event type

Field	Value	Description
EVENT_TYPE_CLOUD_PUBLISH_CDN_STATUS	401	Cloud relay CDN status callback

Event information

Field	Type	Description
RoomId	String/Number	Room ID (Type consistent with client-side room ID type)
RoomType	Number	0 represents numeric room ID, 1 represents string room ID
EventMsTs	Number	Event's Unix timestamp, unit in milliseconds
UserId	String	User ID of the companion robot specified when initiating the task (AgentParams.UserId)
TaskId	Number	Task ID
Payload	JSON Object	Learn more about the event

Payload (Learn more)

Field	Value	Description
Url	String	Push destination URL
Status	Number	Relay status
ErrorCode	Number	Error code
ErrorMsg	String	Error message

Relay status

Field	Value	Description	Callback Frequency
PUBLISH_CDN_STREAM_STATE_IDLE	0	Push not started or ended	Callback only once
PUBLISH_CDN_STREAM_STATE_CONNECTING	1	Connecting TRTC Server and CDN Server	Callback every 5 seconds, no more callbacks after 60 seconds timeout

PUBLISH_CDN_STREAM_STATE_RUNNING	2	CDNs push in progress	Callback only once
PUBLISH_CDN_STREAM_STATE_RECOVERING	3	TRTC server and CDN server push interrupted, recovering	Callback every 5 seconds, no more callbacks after 60 seconds timeout
PUBLISH_CDN_STREAM_STATE_FAILURE	4	TRTC server and CDN server push interrupted, and recovery or connection timeout	Callback only once
PUBLISH_CDN_STREAM_STATE_DISCONNECTING	5	Disconnecting TRTC Server and CDN Server	Callback only once

Relay status recommendation processing

Status	Processing Method
PUBLISH_CDN_STREAM_STATE_IDLE	Indicates URL removal successful, no need to handle.
PUBLISH_CDN_STREAM_STATE_CONNECTING	Indicates URL is connecting, callback every 5s, until connected successfully with `PUBLISH_CDN_STREAM_STATE_RUNNING` , or after 60s callback `PUBLISH_CDN_STREAM_STATE_FAILURE` . You can replace the problematic URL when receiving `PUBLISH_CDN_STREAM_STATE_FAILURE` , and call `UpdatePublishCdnStream` to update Publish parameters. If your business is time-sensitive, you can replace the problematic URL after receiving 2 or more `PUBLISH_CDN_STREAM_STATE_CONNECTING` callbacks, and call `UpdatePublishCdnStream` to update Publish parameters.
PUBLISH_CDN_STREAM_STATE_RUNNING	Indicates URL push successful, no need to handle.

PUBLISH_CDN_STREAM_STATE_RECOVERING	Indicates an interruption occurred during the push process, reconnecting, callback every 5s, until reconnected successfully with `PUBLISH_CDN_STREAM_STATE_RUNNING` , or after 60s callback `PUBLISH_CDN_STREAM_STATE_FAILURE` . Usually caused by network jitter, no need to handle. If `PUBLISH_CDN_STREAM_STATE_RECOVERING` and `PUBLISH_CDN_STREAM_STATE_RUNNING` appear alternately in a short time, you need to check if there are multiple tasks using the same push URL.
PUBLISH_CDN_STREAM_STATE_FAILURE	Indicates push URL connection failed or failed to recover push within 60s, you can replace the problematic URL and call `UpdatePublishCdnStream` to update Publish parameters.
PUBLISH_CDN_STREAM_STATE_DISCONNECTING	Indicates that the push URL is being removed, and after removal successful, it will callback PUBLISH_CDN_STREAM_STATE_IDLE, no need to handle.

Basic Callback Transfer Example

Initiate relay/add relay address to relay success event transfer

`PUBLISH_CDN_STREAM_STATE_CONNECTING` -> `PUBLISH_CDN_STREAM_STATE_RUNNING`

Stop relay/delete relay address to stop relay success event transfer

`PUBLISH_CDN_STREAM_STATE_RUNNING` -> `PUBLISH_CDN_STREAM_STATE_DISCONNECTING` -> `PUBLISH_CDN_STREAM_STATE_IDLE`

During the relay process, connection failure to retry connection success event transfer

`PUBLISH_CDN_STREAM_STATE_RUNNING` -> `PUBLISH_CDN_STREAM_STATE_RECOVERING` -> `PUBLISH_CDN_STREAM_STATE_RUNNING`

During the relay process, connection failure to retry connection timeout failure event transfer

`PUBLISH_CDN_STREAM_STATE_RUNNING` -> `PUBLISH_CDN_STREAM_STATE_RECOVERING` -> `PUBLISH_CDN_STREAM_STATE_FAILURE` -> `PUBLISH_CDN_STREAM_STATE_IDLE`

Note：

Push callback may arrive at your callback server out of order. In this case, you need to sort the events based on the EventMsTs in EventInfo. If you only care about the latest status of the URL, you can ignore the expired events that arrive later.

Signature calculation

Signatures are calculated using the HMAC SHA256 encryption algorithm. Upon receiving a callback message, your server will calculate a signature using the same method, and if the results match, it indicates that the callback is from TRTC and not forged. See below for the calculation method.

```
// In the formula below, `key` is the key used to calculate a signature.
Sign = base64(hmacsha256(key, body))
```

Note:

`body` is the original packet body of the callback request you receive. Do not make any modifications. Below is an example.

```
body="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t103,\\n\\t\\"Callback
```

Verify signature example

Java

Python

PHP

Golang

```
import javax.crypto.Mac;
import javax.crypto.spec.SecretKeySpec;
import java.util.Base64;
//# Function: Third-party callback sign verification
//# Parameters:
//#   key: The key configured in the console
//#   body: The body returned by the Tencent Cloud callback
```

```
//#   sign: The sign value returned by the Tencent Cloud callback
//# Return Value:
//#   Status: OK indicates that the verification has passed, FAIL indicates that th
//#   Info: Success/Failure information

public class checkSign {
    public static String getResultSign(String key, String body) throws Exception {
        Mac hmacSha256 = Mac.getInstance("HmacSHA256");
        SecretKeySpec secret_key = new SecretKeySpec(key.getBytes(), "HmacSHA256");
        hmacSha256.init(secret_key);
        return Base64.getEncoder().encodeToString(hmacSha256.doFinal(body.getBytes(
    }
    public static void main(String[] args) throws Exception {
        String key = "123654";
        String body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\
        String Sign = "kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA=";
        String resultSign = getResultSign(key, body);

        if (resultSign.equals(Sign)) {
            System.out.println("{'Status': 'OK', 'Info': 'validation passed'}");
        } else {
            System.out.println("{'Status': 'FAIL', 'Info': 'validation failed'}");
        }
    }
}
```



```
# -*- coding: utf8 -*-
import hmac
import base64
from hashlib import sha256

# Function: Third-party callback sign verification
# Parameters:
#   key: The key configured in the console
#   body: The body returned by the Tencent Cloud callback
#   sign: The sign value returned by the Tencent Cloud callback
# Return Value:
```

```
#   Status: OK indicates that the verification has passed, FAIL indicates that the
#   Info: Success/Failure information

def checkSign(key, body, sign):
    temp_dict = {}
    computSign = base64.b64encode(hmac.new(key.encode('utf-8'), body.encode('utf-8'
    print(computSign)
    if computSign == sign:
        temp_dict['Status'] = 'OK'
        temp_dict['Info'] = 'validation passed'
        return temp_dict
    else:
        temp_dict['Status'] = 'FAIL'
        temp_dict['Info'] = 'validation failed'
        return temp_dict

if __name__ == '__main__':
    key = '123654'
    body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\\":\\t204,\
    sign = 'kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA='
    result = checkSign(key, body, sign)
    print(result)
```



```
<?php

class TlsEventSig {

    private $key = false;
    private $body = false;

    public function __construct( $key, $body ) {
        $this->key = $key;
                $this->body = $body;
    }
```

```
    private function __hmacsha256() {
        $hash = hash_hmac( 'sha256', $this->body, $this->key, true );
                return base64_encode( $hash);
    }

    public function genEventSig() {
        return $this->__hmacsha256();
    }
}

$key="789";
$data="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\\"Callbac

$api = new  TlsEventSig($key, $data);
echo $api->genEventSig();
```

```
package main
import "fmt"
import (
        "crypto/hmac"
        "crypto/sha256"
        "encoding/base64"
)

func main () {
    var data = "{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\
    var key = "789"
```

```
    //JSRUN engine 2.0, supporting up to 30 types of languages for online running,
    fmt.Println(hmacsha256(data,key))
}

func hmacsha256(data string, key string) string {
        h := hmac.New(sha256.New, []byte(key))
        h.Write([]byte(data))
        return base64.StdEncoding.EncodeToString(h.Sum(nil))
}
```

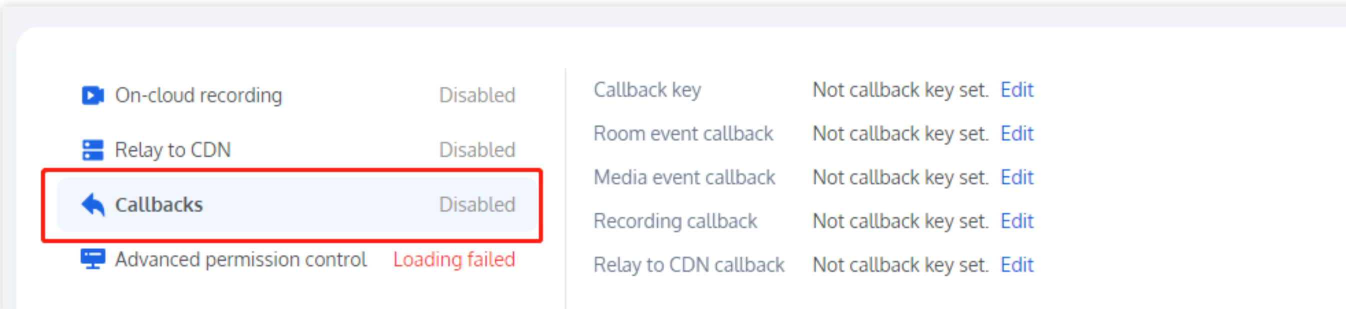
Cloud Recording Callback

Last updated：2024-08-07 10:53:53

This document describes the callback events of the updated cloud recording feature.

Configuration Information

On the Tencent RTC Console, you can configure callback information. Upon configuration completion, you can receive event callback notifications.



Note：

You need to prepare the following information in advance and complete the Callback Configuration in the console.

Required: the HTTP/HTTPS server address to receive callback notifications.

Optional: Key for signature computation. You can customize a key of up to 32 characters, composed of uppercase and lowercase letters and numbers.

Timeout Retry

If your server does not respond within 5 seconds after the event callback server sends the message notification, it is deemed as a failed notification. If the initial notification fails, an immediate retry is performed. If the notification fails again, retry will be performed at an interval of **10 seconds** until the message has been kept for 1 minute, after which retries will not be performed.

Callback API

You can assign an HTTP/HTTPS service gateway for subscribing to callback messages. When any related event happens, the cloud recording system calls back the event notification to your message receiving server.

Format of the Event Callback Message

Event callback messages are sent to your server via HTTP/HTTPS POST requests, in which:

Character Encoding Format: UTF-8.

Request: The body is in the JSON format.

Response: HTTP STATUS CODE = 200. The server ignores the specific content of the response package. To ensure protocol friendliness, it is recommended that the customer put the following in the response content: JSON: {"code":0}.

Parameter Description

The Header of the Event Callback Message Contains the Following Fields:

Field name	Value
Content-Type	application/json
Sign	Signature value
SdkAppId	sdk application id

The Body of the Event Callback Message Includes the Following Fields:

Field Name	Type	Description
EventGroupId	Number	Event group ID, fixed at 3 for cloud recording
EventType	Number	Event type for callback notification
CallbackTs	Number	The Unix timestamp (in milliseconds) when the event callback server sends a callback request to your server
EventInfo	JSON Object	The event information

Event Type Description:

Field Name	Type	Description
EVENT_TYPE_CLOUD_RECORDING_RECORDER_START	301	The cloud recording module starts.
EVENT_TYPE_CLOUD_RECORDING_RECORDER_STOP	302	The cloud recording module exits.
EVENT_TYPE_CLOUD_RECORDING_UPLOAD_START	303	The cloud recording file upload task starts, and it

		is called back only when COS is chosen.
EVENT_TYPE_CLOUD_RECORDING_FILE_INFO	304	Cloud recording: Generating the M3U8 index file. After the first successful generation and upload, it is called back only when COS is chosen through API recording.
EVENT_TYPE_CLOUD_RECORDING_UPLOAD_STOP	305	The cloud recording file upload is complete. It is called back only when COS is chosen.
EVENT_TYPE_CLOUD_RECORDING_FAILOVER	306	Cloud recording migration occurs. It is triggered when the existing recording task is migrated to a new load.
EVENT_TYPE_CLOUD_RECORDING_FILE_SLICE	307	Cloud recording: Generating the M3U8 index file (generating the first ts slice). After generation, it is called back only when COS is chosen through API recording.
EVENT_TYPE_CLOUD_RECORDING_DOWNLOAD_IMAGE_ERROR	309	An error occurs when the cloud recording attempts to download and decode the image file.
EVENT_TYPE_CLOUD_RECORDING_MP4_STOP	310	The MP4 recording task of cloud recording stops, and it is called back only when COS is chosen via API recording (when automatic recording is turned on in the console and the authorized VOD COS is selected as

		storage, please pay attention to event 311).
EVENT_TYPE_CLOUD_RECORDING_VOD_COMMIT	311	The cloud recording VOD recording task has completed the upload of media resources, and it is called back when you select **Video on Demand** and **automatically recording to COS** via the console (after file recording ends, the VOD index information is carried. Please subscribe to this type of callback event).
EVENT_TYPE_CLOUD_RECORDING_VOD_STOP	312	The cloud recording VOD task stops, and it is called back only when VOD is chosen.

Note：

The callback statuses from event types 301 to 309 are intermediate states of real-time recording, for you to better understand the recording process and keep track of the status. The successful upload of the actual recording file to video on demand will trigger an event 311 callback, and the overall task is completed and an event 312 is called back.

Event Information Description:

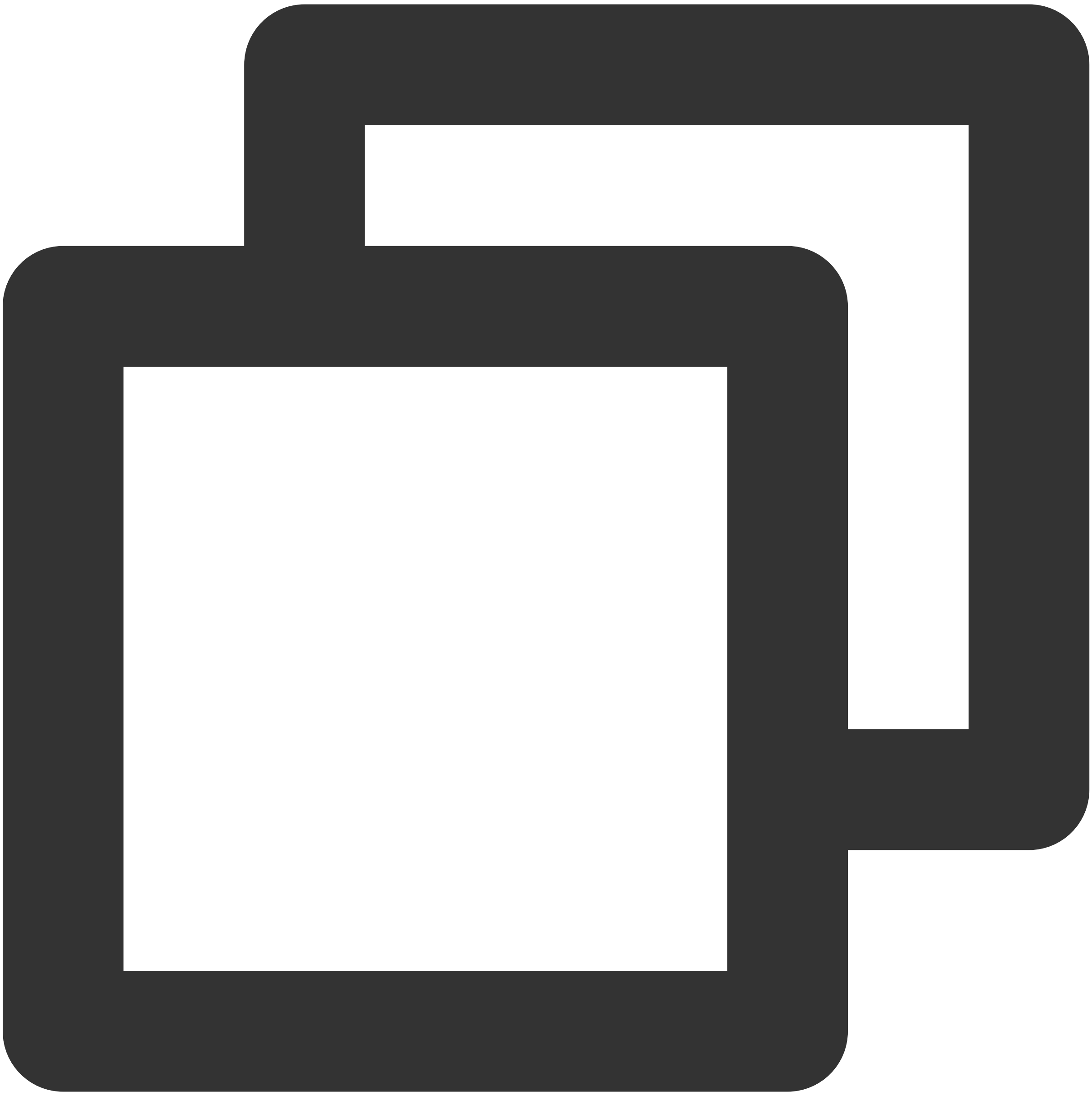
Field Name	Type	Description
RoomId	String/Number	The room name, consistent with the room ID type on the client side
EventTs	Number	The Unix timestamp of when the event occurred, in seconds (this field is not recommended. Instead, EventMsTs is recommended)
EventMsTs	Number	The Unix timestamp of when the event occurred, in milliseconds
UserId	String	The user ID of the recording robot
TaskId	String	The recording ID, which is a unique ID of a single cloud recording task
Payload	JsonObject	Defined based on various event types

When the event type is

301

(EVENT_TYPE_CLOUD_RECORDING_RECORDER_START), the definition of Payload is as follows:

Field Name	Type	Description
Status	Number	0: Indicates that the recording module successfully starts. 1: Indicates that the recording module fails to start.



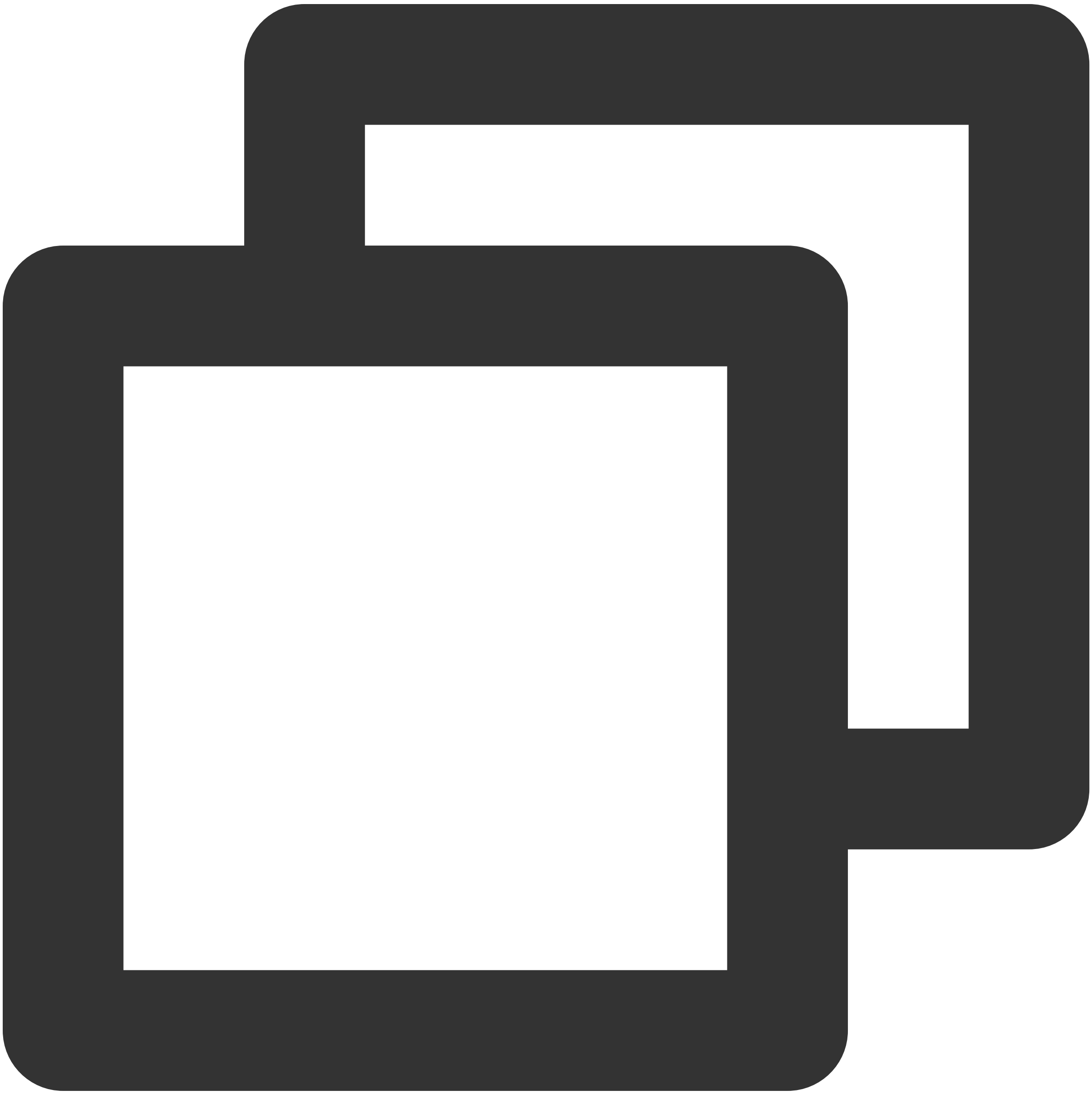
```
{
    "EventGroupId": 3,
    "EventType": 301,
    "CallbackTs": 1622186275913,
    "EventInfo": {
        "RoomId": "xx",
        "EventTs": "1622186275",
        "EventMsTs": 1622186275757,
        "UserId": "xx",
        "TaskId": "xx",
        "Payload": {
            "Status": 0
        }
    }
}
```

When the event type is

302

(EVENT_TYPE_CLOUD_RECORDING_RECORDER_STOP), the definition of Payload is as follows:

Field Name	Type	Description
LeaveCode	Number	0: The invocation of recording module normally stops and exits. 1: The customer kicks out the recording robot from the room. 2: The customer disbands the room. 3: The server kicks out the recording robot from the room. 4: The server disbands the room. 99: There is no other user flow in the room except for the recording robot, which will exit after a specified time. 100: Exit from the room due to timeout. 101: Repeated entry of the same user into the same room causes the robot to exit.



```
{
  "EventGroupId": 3,
  "EventType": 302,
  "CallbackTs": 1622186354806,
  "EventInfo": {
    "RoomId": "xx",
    "EventTs": "1622186354",
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "LeaveCode": 0
      }
    }
  }
```

When the event type is
303
(EVENT_TYPE_CLOUD_RECORDING_UPLOAD_START), the definition of Payload is as follows:

Field Name	Type	Description
Status	Number	0: Indicates that the upload module normally starts. 1: Indicates that the upload module fails to be initiated.

When the event type is
304
(EVENT_TYPE_CLOUD_RECORDING_FILE_INFO), the definition of Payload is as follows:

Field Name	Type	Description
FileList	String	The generated M3U8 filename

When the event type is
305
(EVENT_TYPE_CLOUD_RECORDING_UPLOAD_STOP), the definition of Payload is as follows:

Field Name	Type	Description
LeaveCode	Number	0: Indicates that this recording upload task has been completed, and all files have been uploaded to the specified third-party cloud storage. 1: Indicates that this recording upload task has been completed, but at least one file is lingering on the server or backup storage. 2: Indicates that files lingering on the server or backup storage have been restored and uploaded to the designated third-party cloud storage. Note: 305 indicates the event that the HLS file upload is complete.

When the event type is
306
(EVENT_TYPE_CLOUD_RECORDING_FAILOVER), the definition of Payload is as follows:

Field Name	Type	Description
Status	Number	0: Indicates that this migration is completed.

```
{
  "EventGroupId": 3,
  "EventType": 306,
  "CallbackTs": 1622191989674,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191989,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
```

```
      "TaskId": "xx",
      "Payload": {
        "Status": 0
      }
    }
  }
```

When the event type is

307

(EVENT_TYPE_CLOUD_RECORDING_FILE_SLICE), the definition of Payload is as follows:

Field Name	Type	Description
FileName	String	M3U8 filename
UserId	String	The user ID corresponding to the recorded file
TrackType	String	Audio/Video types: audio/video/audio_video
BeginTimeStamp	String	The Unix timestamp of the server when the recording starts (in milliseconds)

When the event type is

309

(EVENT_TYPE_CLOUD_RECORDING_DOWNLOAD_IMAGE_ERROR), the definition of Payload is as follows:

Field Name	Type	Description
Url	String	The URL of the failed download

```
{
  "EventGroupId": 3,
  "EventType": 309,
  "CallbackTs": 1622191989674,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191989,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "Url": "http://xx"
      }
    }
  }
```

When the event type is

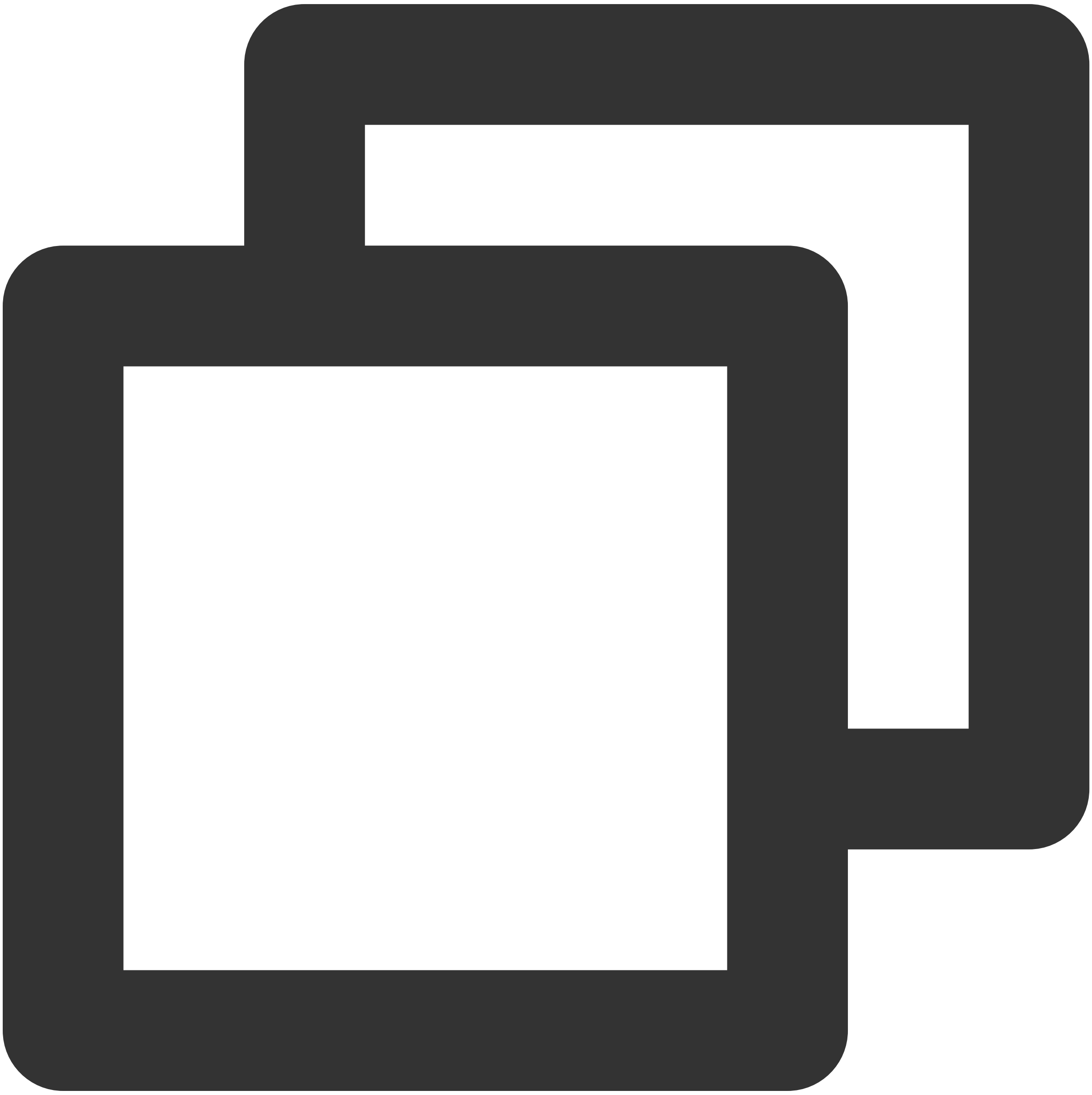
310

(EVENT_TYPE_CLOUD_RECORDING_MP4_STOP), the definition of Payload is as follows:

Note：

310 is a callback event after an MP4 file is uploaded to the user-specified third-party COS. A recording task may call back multiple events of 310 (each event corresponds to a recorded file information).

Field Name	Type	Description
Status	Number	0: Indicates that the MP4 recording task has exited normally, and all files have been uploaded to the designated third-party cloud storage. 1: Indicates that this MP4 recording task has exited normally, but at least one file lingers on the server or backup storage. 2: Indicates that this MP4 recording task exits abnormally (the possible reason is the failure of extracting HLS files from COS).
FileList	Array	All generated MP4 file names
FileMessage	Array	All generated MP4 file information
FileName	String	MP4 file name
UserId	String	The user ID corresponding to the MP4 file (this field is empty when the recording mode is set to mixed streaming mode)
TrackType	String	audio for audio / video for pure video / audio_video for audio and video
MediaId	String	The primary and auxiliary stream tag. main indicates the primary stream (camera), aux indicates the auxiliary streams (screen sharing), and mix indicates mixed stream recording.
StartTimeStamp	Number	The Unix timestamp at the beginning of the MP4 file (in milliseconds)
EndTimeStamp	Number	The UNIX timestamp at the end of the MP4 file (in milliseconds)




```
{
  "EventGroupId": 3,
  "EventType": 310,
  "CallbackTs": 1622191965320,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191989,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "Status": 0,
        "FileList": ["xxxx1.mp4", "xxxx2.mp4"],
        "FileMessage": [
          {
              "FileName": "xxxx1.mp4",
              "UserId": "xxxx",
              "TrackType": "audio_video",
              "MediaId": "main",
              "StartTimeStamp": 1622186279145,
              "EndTimeStamp": 1622186282145
          },
          {
              "FileName": "xxxx2.mp4",
              "UserId": "xxxx",
              "TrackType": "audio_video",
              "MediaId": "main",
              "StartTimeStamp": 1622186279153,
              "EndTimeStamp": 1622186282153
          }
       ]

     }
   }
 }
```


When the event type is

311

(EVENT_TYPE_CLOUD_RECORDING_VOD_COMMIT), the definition of Payload is as follows:

Field Name	Type	Description
Status	Number	0: Indicates that the recorded file has been successfully uploaded to the VOD platform. 1: Indicates that the recorded file lingers on the server or backup storage. 2: Indicates that the upload and VOD task for this recorded file to is abnormal.
UserId	String	The user ID corresponding to this recorded file (this field is empty when the recording mode is set to mixed stream mode)
TrackType	String	audio for audio / video for pure video / audio_video for audio and video
MediaId	String	The primary and auxiliary stream tag. main indicates the primary stream (camera), aux indicates the auxiliary stream (screen sharing), and mix indicates mixed stream recording.

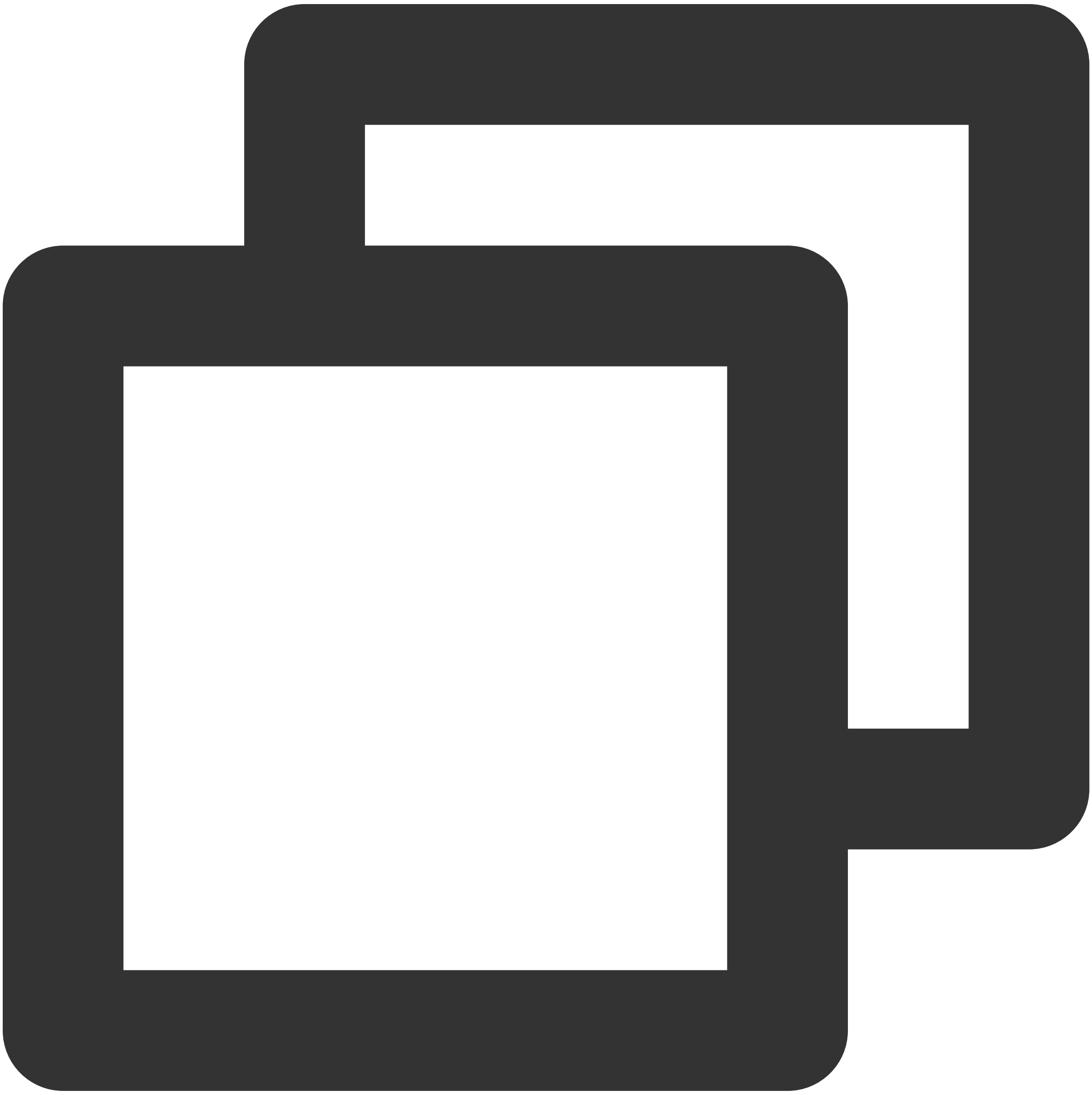
FileId	String	The unique ID of this recorded file in the VOD platform
VideoUrl	String	The playback address of this recorded file on the VOD platform
CacheFile	String	The filename corresponding to this MP4/HLS recording file
StartTimeStamp	Number	The Unix timestamp of the beginning of this recorded file (in milliseconds)
EndTimeStamp	Number	The Unix timestamp of the end of this recorded file (in milliseconds)
Errmsg	String	Corresponding error message when the status is not 0

Callback for successful upload:

```
{
  "EventGroupId": 3,
  "EventType": 311,
  "CallbackTs": 1622191965320,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191965,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "Status": 0,
        "TencentVod": {
          "UserId": "xx",
          "TrackType": "audio_video",
          "MediaId": "main",
          "FileId": "xxxx",
          "VideoUrl": "http://xxxx",
          "CacheFile": "xxxx.mp4",
          "StartTimeStamp": 1622186279153,
          "EndTimeStamp": 1622186282153
        }
      }
    }
  }
```

Callback for upload failure:




```
{
  "EventGroupId": 3,
  "EventType": 311,
  "CallbackTs": 1622191965320,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191965,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "Status": 1,
        "Errmsg": "xxx",
        "TencentVod": {
          "UserId": "123",
          "TrackType": "audio_video",
          "CacheFile": "xxx.mp4"
        }
      }
    }
  }
```

Note：

After the callback of 311 is received, the **file upload is completed,** and you need to wait for 30 seconds to 3 minutes for the file to be completely recorded, depending on the file size.

When the event type is

312

(EVENT_TYPE_CLOUD_RECORDING_VOD_STOP), the definition of Payload is as follows:

Field Name	Type	Description
Status	Number	0: Indicates that this VOD upload task has exited normally. 1: Indicates that this VOD upload task exits abnormally.

```
{
  "EventGroupId": 3,
  "EventType": 312,
  "CallbackTs": 1622191965320,
  "EventInfo": {
    "RoomId": "20015",
    "EventTs": 1622191965,
    "EventMsTs": 1622186275757,
    "UserId": "xx",
    "TaskId": "xx",
    "Payload": {
```

```
        "Status": 0
      }
    }
  }
```

Calculating a Signature

Signatures are calculated using the HMAC SHA256 encryption algorithm. After your event callback server receives the callback message, it calculates the signature in the same manner. If they match, it is an event callback from Tencent Real-Time Communication (TRTC), not a forged one. The signature calculation is as follows:

```
// In the signature Sign calculation formula, key is the encryption key used for ca
Sign = base64(hmacsha256(key, body))
```

Note：

The body is the original package body of the callback request received by you. Do not convert it. See the following example:

```
body="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t103,\\n\\t\\"Callback
```

Signature Verification Example

Java

Python

PHP

Golang

```
import javax.crypto.Mac;
import javax.crypto.spec.SecretKeySpec;
import java.util.Base64;
//# Function: Third-party callback sign verification
```

```
//# Parameters:
//#   key: The key configured in the console
//#   body: The body returned in Tencent Cloud callback
//#   sign: Signature value of sign returned in Tencent Cloud callback
//# Returned values:
//#   Status: OK indicates successful verification, and FAIL indicates verification
//#   Info: Success/Failure message

public class checkSign {
    public static String secureFinalSign(String key, String entityBody) throws Exce
        Mac hmacSha256 = Mac.getInstance("HmacSHA256");
        SecretKeySpec secret_key = new SecretKeySpec(key.getBytes(), "HmacSHA256");
        hmacSha256.initialize(secret_key);
        return Base64.getEncoder().encodeToString(hmacSha256.doFinal(body.getBytes(
    }
    public static void main(String[] args) throws Exception {
        String key = "123654";
        String body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\
        String Sign = "kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA=";
        String resultSign = obtainResultSignature(key, body);

        if (resultSign.equals(Sign)) {
            System.out.println("{'Status': 'OK', 'Info': 'Verification passed'}");
        } else {
            System.out.println("{'Status': 'FAIL', 'Info': 'Verification failed'}")
        }
    }
}
```



```
# -*- coding: utf8 -*-
import hmac
import base64
from hashlib import sha256

# Function: Third-party callback sign verification
# Parameters:
#   key: The key configured in the console
#   body: The body returned in Tencent Cloud callback
#   Sign: The signature value sign returned in Tencent Cloud's callback
# Returned values:
```

```
# Status: OK indicates successful verification, and FAIL indicates verification fai
#   Info: Success/Failure Information

def checkSign(key, body, sign):
    temp_dict = {}
    computSign = base64.b64encode(hmac.new(key.encode('utf-8'), body.encode('utf-8'
    print(computSign)
    if computSign equals sign:
        temp_dict['Status'] = 'OK'
        temp_dict['Info'] = 'Verification passed'
        return temp_dict
    else:
        temp_dict['Status'] = 'FAIL'
        temp_dict['Info'] = 'Verification failed'
        return temp_dict

if __name__ == '__main__':
    key = '123654'
    body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\\":\\t204,\
    `sign` = 'kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA='
    result = verifySignature(key, body, sign)
    print(result)
```



```
<?php

class TlsEventSig {

    private $key = false;
    private $body = false;

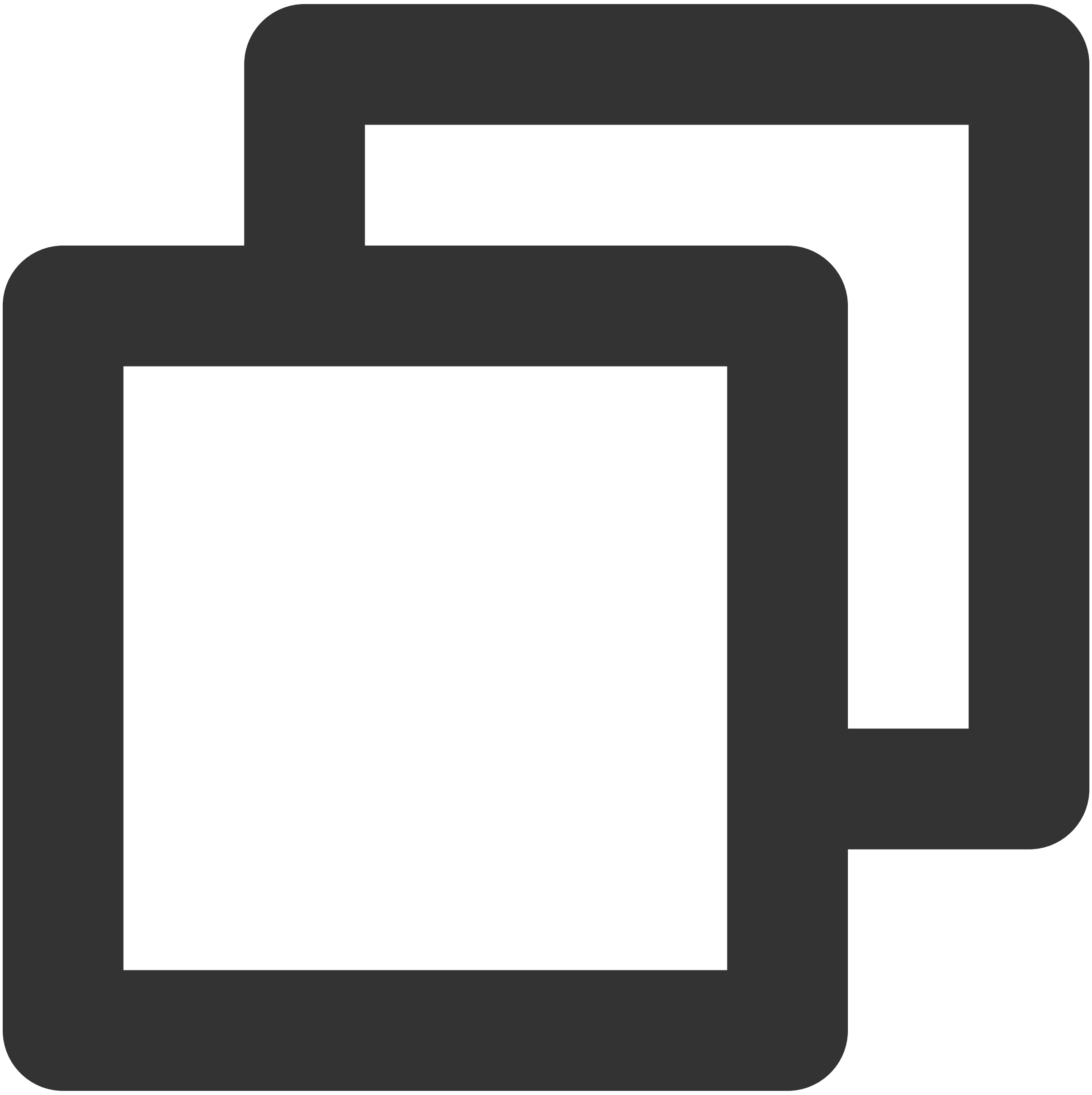
    public function __construct( $key, $body ) {
        $this->key = $key;
                $this->body = $body;
    }
```

```
    private function __hmacsha256() {
        $hash = hash_hmac( 'sha256', $this->body, $this->key, true );
                return base64_encode( $hash);
    }

    public function genEventSig() {
        return $this->__hmacsha256();
    }
}

$key="789";
$data="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\\"Callbac

$api = new  TlsEventSig($key, $data);
echo $api->genEventSig();
```



```
package main
import "fmt"
import (
        "crypto/hmac"
        "crypto/sha256"
        "encoding/base64"
)

func main () {
    var data = "{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\
    var key = "789"
```

```
    //JSRUN engine 2.0, supporting up to 30 types of languages for online running,
    fmt.Println(hmacsha256(data,key))
}

func hmacsha256(data string, key string) string {
        h := hmac.New(sha256.New, []byte(key))
        h.Write([]byte(data))
        return base64.StdEncoding.EncodeToString(h.Sum(nil))
}
```

Push Online Media Stream Callback

Last updated：2024-08-01 16:20:02

The server-side push online media stream callback supports notifying your server of push online media stream events generated using the Push Online Media Stream REST API in the form of HTTP/HTTPS requests. You can provide Tencent Cloud with relevant configuration information to enable this service.

Configuration Information

The TRTC console supports self-service configuration of callback information. Once the configuration is completed, you can receive event callback notifications. For detailed operation instructions, see the Callback Configuration.

Note:

You need to prepare the following information in advance:

Required item: An HTTP/HTTPS server address to receive callback notifications.

Optional item: A key for signature calculation, which is a key of up to 32 characters defined by you and consists of uppercase and lowercase letters and numbers.

Timeout Retry

If the event callback server does not receive a response from your server within 5 seconds after sending a message notification, the notification is considered to have failed. After the first notification failure, a retry will be made immediately. Subsequent retries will occur at 10-second intervals until the message retention time exceeds 1 minute, after which no further retries will be made.

Format of Event Callback Messages

Event callback messages are sent to your server via HTTP/HTTPS POST requests, where:

Character Encoding Format: UTF-8.

Request: The body is in the JSON format.

Response: HTTP STATUS CODE = 200. The server ignores the specific content of the response package. For the sake of protocol friendliness, it is recommended that the customer response carry JSON: {"code":0}.

Package Body Example: Below is a package body example for the "push online media stream started successfully" event.


```
}
```


```
{
        "EventGroupId": 7,
        "EventType":    701,
        "CallbackMsTs":   1701937900012,
        "EventInfo":    {
                "EventMsTs": 1701937900013,
                "TaskId":"xx",
                "Status":0
        }
}
```


Description of Parameters

Callback Message Parameters

The header of the event callback message contains the following fields:

Field name	Value
Content-Type	application/json
Sign	Signature value
SdkAppId	sdk application id

The body of the event callback message includes the following fields:

Field name	Type	Meaning
EventGroupId	Number	Event group ID, which is 4 for a stream mixing and relaying event
EventType	Number	The event type of the callback notification
CallbackMsTs	Number	The Unix timestamp of the callback request sent by the event callback server to your server, in milliseconds
EventInfo	JSON Object	Event information

Event Group ID

Field name	Value	Meaning
EVENT_GROUP_STREAM_INGEST	7	Push online media stream event group

Event Type

Field name	Value	Meaning
EVENT_TYPE_STREAM_INGEST_START	701	Push online media stream start
EVENT_TYPE_STREAM_INGEST_STOP	702	Push online media stream stop

Definition of Event Information when the Event Type is (EVENT_TYPE_STREAM_INGEST_START 701):

Field name	Type	Meaning

EventMsTs	String	The Unix timestamp of the event occurred, in milliseconds
TaskId	String	Push online media stream task ID
Status	Number	Status of push online media stream start

Status of Push Online Media Stream Start

Field name	Value	Meaning	Callback frequency
STATUS_START_SUCCESS	0	The push online media stream start succeeded.	Callback is made once upon success.
STATUS_START_FAILURE	1	The push online media stream start failed.	Callback is made once upon failure.
STATUS_START_AGAIN	2	The push online media stream starts again.	A retry is made at the 0th, 1st, and 3rd second, with callback during the retry.

Recommended Handling of Push Online Media Stream Status

Status	Handling method
STATUS_START_SUCCESS	It indicates success, with no need for handling.
STATUS_START_FAILURE	If you receive push online media stream failure status three times, check the source URL and restart the push online media stream.
STATUS_START_AGAIN	Received within 1 minute after the push online media stream starts: It indicates the URL connection failed or the RTMP push failed. The system automatically triggers a retry. If it fails in the end, check if the URL is properly connected Received beyond 1 minute after the push online media stream starts: A restart may be triggered due to source stream or background network fluctuation, with no need for handling.

Basic Callback Transfer Example

Event transfer of push online media stream failure/push online media stream restart/push online media stream start success

`STATUS_START_FAILURE` -> `STATUS_START_AGAIN` -> `STATUS_START_SUCCESS`

Note:

Push online media stream callback events may arrive at your callback server out of sequence. You need to sort events based on EventMsTs in EventInfo. If you only care about the latest status of the URL, you can ignore the expired events that arrive later.

Definition of Event Information when the Event Type is (EVENT_TYPE_STREAM_INGEST_STOP 702):

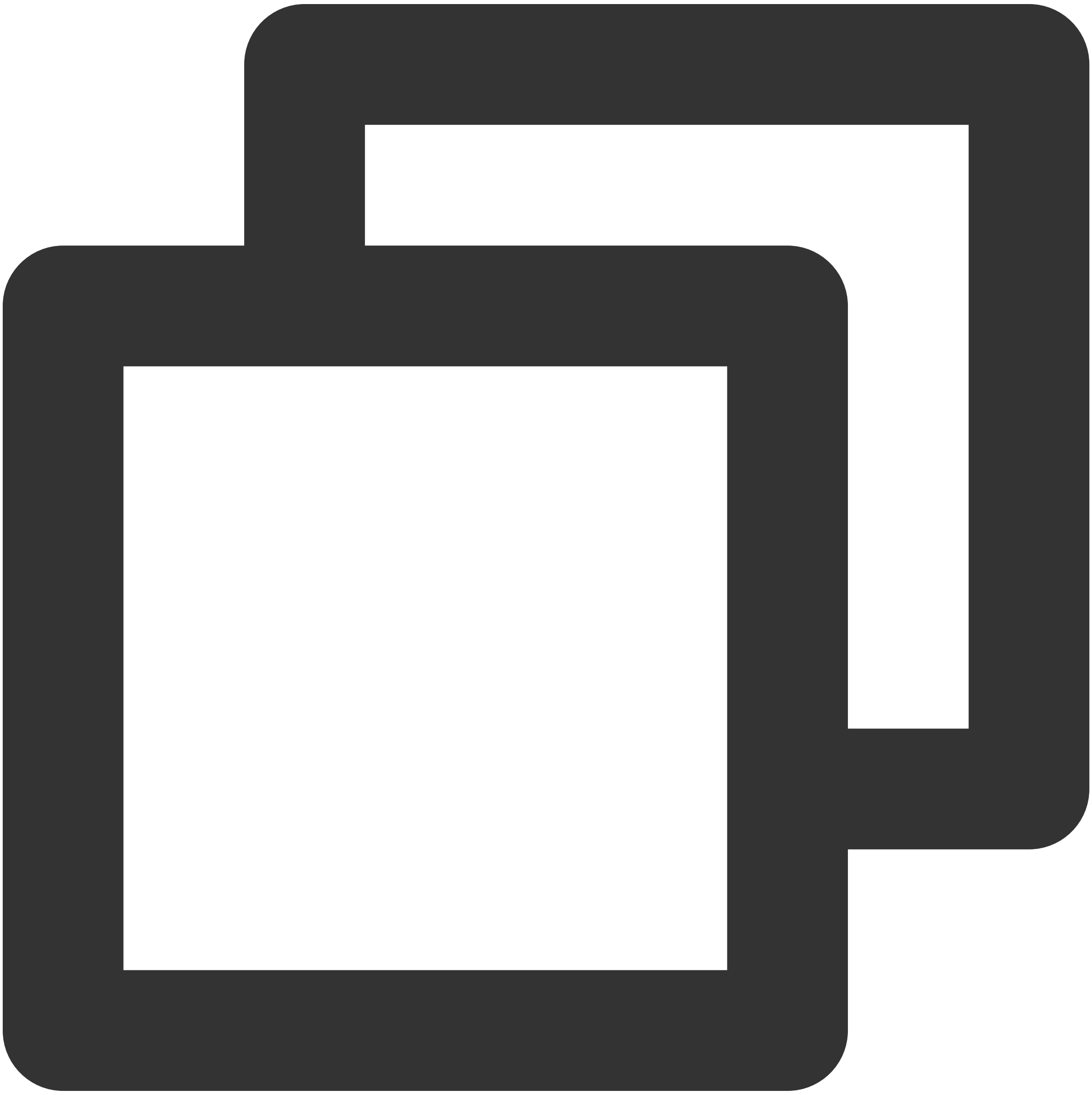
Field name	Type	Meaning
EventMsTs	String	The Unix timestamp of the event occurred, in milliseconds
TaskId	String	Push online media stream task ID
Status	Number	Status of push online media stream stop

Status of Push Online Media Stream Stop

Field name	Value	Meaning	Callback Frequency
STATUS_STOP_SUCCESS	0	The push online media stream stop succeeded.	Callback is made once upon success.

Calculating a Signature

A signature is calculated with the HMAC SHA256 encryption algorithm. After your event callback server receives the callback message, it calculates the signature in the same manner. A match means that it is TRTC's event callback, with no falsification. The signature calculation is shown below:



```
// In the signature calculation formula Sign, the key refers to the encryption key
Sign = base64(hmacsha256(key, body))
```

Note:

The body refers to the original package body of the callback request received by you, with no transformation. An example is as follows:


```
body="{\\n\\t\\"Ebody="{\\"EventGroupId\\":7,\\"EventType\\":701,\\"CallbackMsTs\\"
```

Signature Verification Example

Java

Python

PHP

Golang

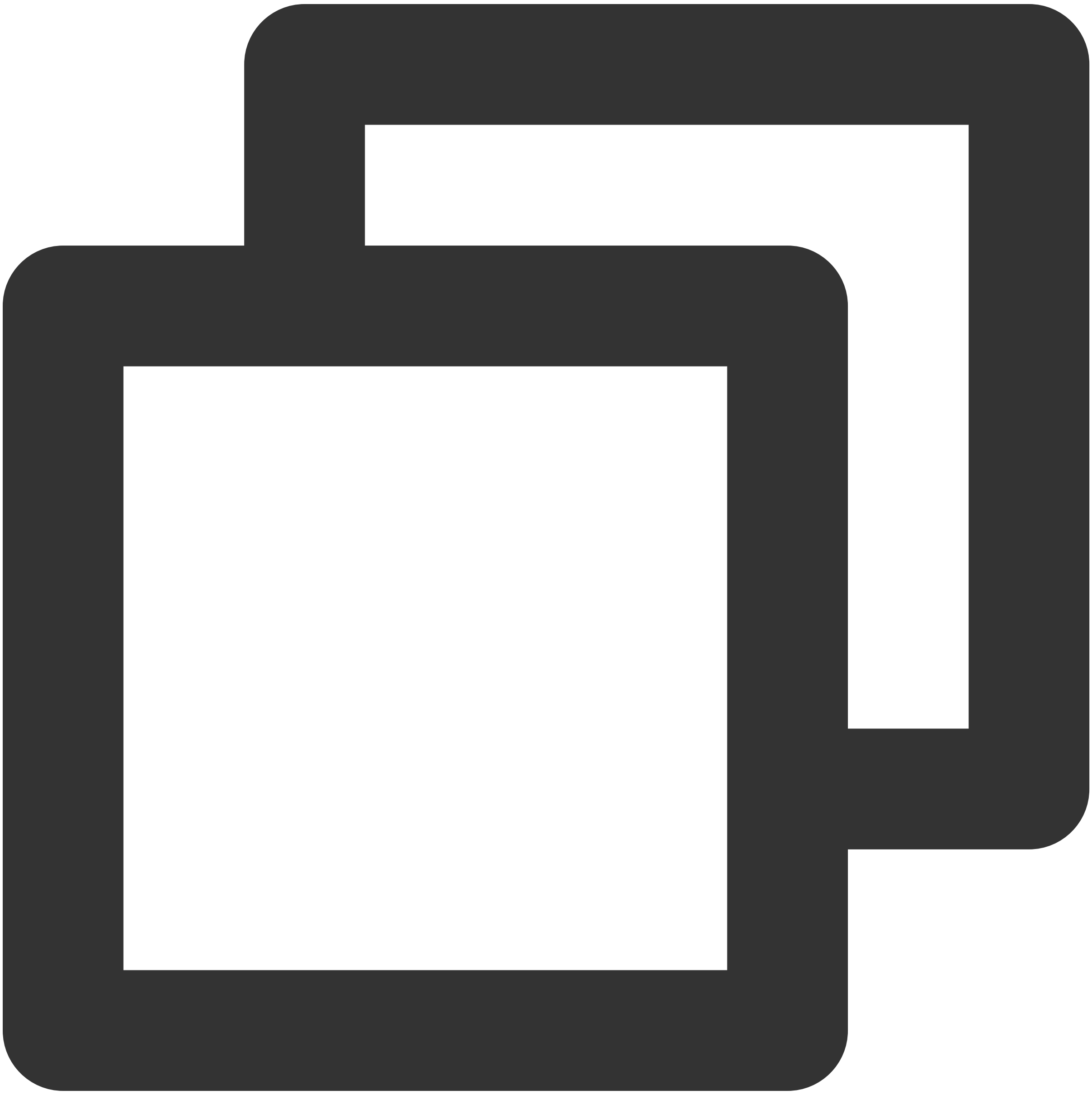
```
import javax.crypto.Mac;
import javax.crypto.spec.SecretKeySpec;
import java.util.Base64;
//# Feature: Third-party Callback Sign Verification
//# Parameters:
//#   key: The Key Configured on the Console
//#   body: The Body Returned by Tencent Cloud Callback
//#   sign: The Signature Value Returned by Tencent Cloud Callback
//# Returned Values:
```



```
//#   Status: OK Indicates that Verification Succeeded, and FAIL Indicates that Ver
//#   Info: Success/Failure Information

public class checkSign {
    public static String secureFinalSign(String key, String entityBody) throws Exce
        Mac hmacSha256 = Mac.getInstance("HmacSHA256");
        SecretKeySpec secret_key = new SecretKeySpec(key.getBytes(), "HmacSHA256");
        hmacSha256.initialize(secret_key);
        return Base64.getEncoder().encodeToString(hmacSha256.doFinal(body.getBytes(
    }
    public static void main(String[] args) throws Exception {
        String key = "123654";
        String body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\
        String Sign = "kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA=";
        String resultSign = obtainResultSignature(key, body);

        if (resultSign.equals(Sign)) {
            System.out.println("{'Status': 'OK', 'Info': 'Verification succeeded'}"
        } else {
            System.out.println("{'Status': 'FAIL', 'Info': 'Verification failed'}")
        }
    }
}
```



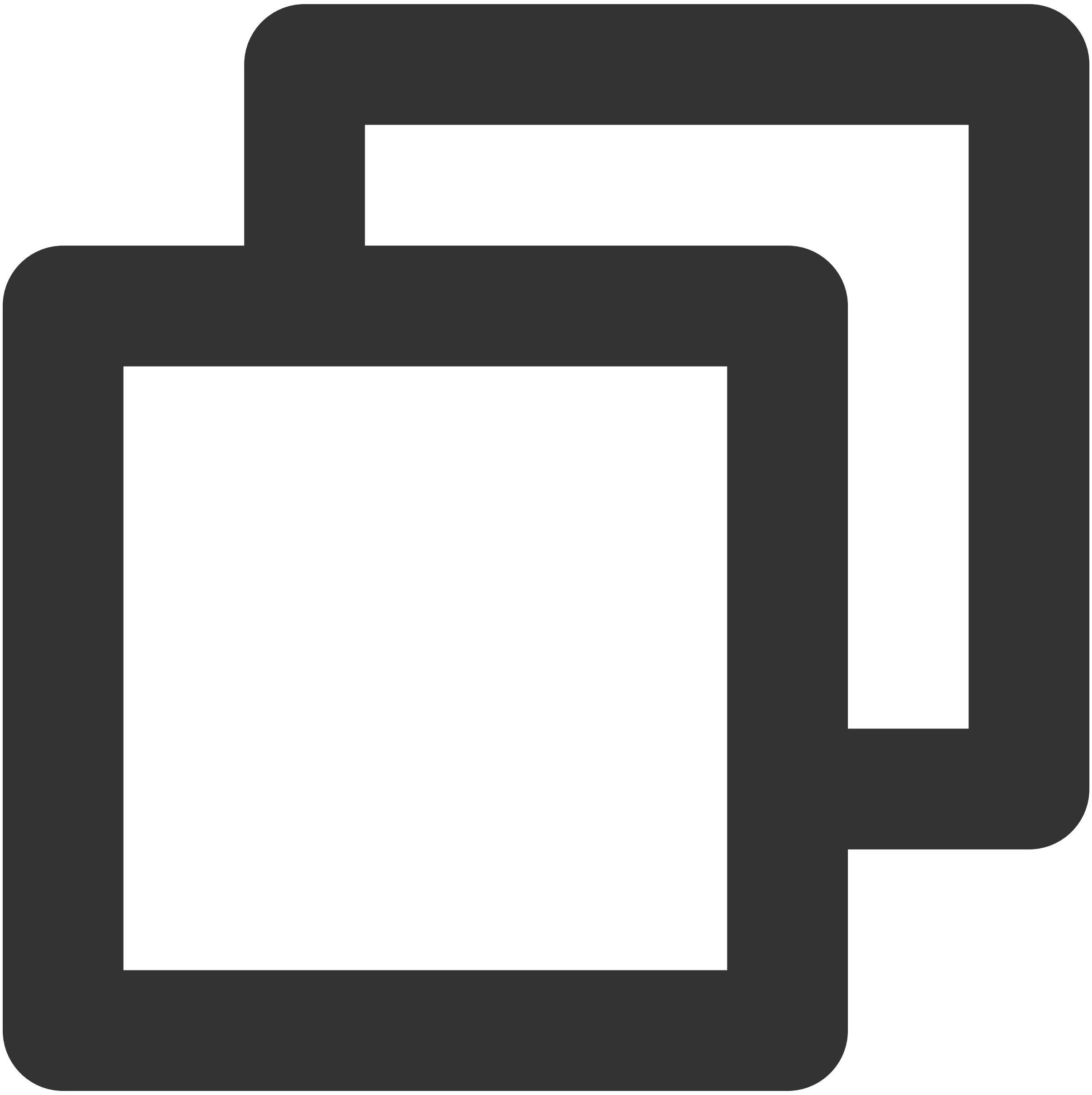
```
# -*- coding: utf8 -*-
import hmac
import base64
from hashlib import sha256

# Feature: Third-party Callback Sign Verification
# Parameters:
#   key: The Key on the Console
#   body: The Body Returned by Tencent Cloud Callback
#   sign: The Signature Value Returned by Tencent Cloud Callback
# Returned Values:
```

```
#   Status: OK Indicates that Verification Succeeded, and FAIL Indicates that Verif
#   Info: Success/Failure Information

def checkSign(key, body, sign):
    temp_dict = {}
    computSign = base64.b64encode(hmac.new(key.encode('utf-8'), body.encode('utf-8'
    print(computSign)
    if computSign equals sign:
        temp_dict['Status'] = 'OK'
        temp_dict['Info'] = 'Verification succeeded'
        return temporary_dictionary
    else:
        temp_dict['Status'] = 'FAIL'
        temp_dict['Info'] = 'Verification failed'
        return temporary_dictionary

if __name__ == '__main__':
    key = '123654'
    body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\\":\\t204,\
    `sign` = 'kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA='
    result = verifySignature(key, body, sign)
    print(result)
```



```
<?php

class TlsEventSig {

    private $key = false;
    private $body = false;

    public function __construct( $key, $body ) {
        $this->key = $key;
                $this->body = $body;
    }
```

```
    private function __hmacsha256() {
        $hash = hash_hmac( 'sha256', $this->body, $this->key, true );
                return base64_encode( $hash);
    }

    public function genEventSig() {
        return $this->__hmacsha256();
    }
}

$key="789";
$data="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\\"Callbac

$api = new  TlsEventSig($key, $data);
echo $api->genEventSig();
```

```
package main
import "fmt"
import (
        "crypto/hmac"
        "crypto/sha256"
        "encoding/base64"
)

func main () {
    var data = "{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\
    var key = "789"
```

```
    //JSRUN Engine 2.0, which supports online running in up to 30 languages and ful
    fmt.Println(hmacsha256(data,key))
}

func hmacsha256(data string, key string) string {
        h := hmac.New(sha256.New, []byte(key))
        h.Write([]byte(data))
        return base64.StdEncoding.EncodeToString(h.Sum(nil))
}
```

Verify Signature Example

Last updated：2023-10-08 16:01:32

Tencent Real-Time Communication(TRTC) console supports self-configuration of callback information. After the configuration is completed, you can receive event callback notifications. Before configuring the callback information, you need to prepare a key for signature calculation. You can define a key with a maximum of 32 characters, composed of uppercase and lowercase letters and numbers.
This document will help you verify signature after calculating it and show you how to perform an example.

Signature calculation

Signatures are calculated using the HMAC SHA256 encryption algorithm. Upon receiving a callback message, your server will calculate a signature using the same method, and if the results match, it indicates that the callback is from TRTC and not forged. See below for the calculation method.

```
// In the formula below, `key` is the key used to calculate a signature.
Sign = base64(hmacsha256(key, body))
```

Note

`body` is the original packet body of the callback request you receive. Do not make any modifications. Below is an example.

```
body="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t103,\\n\\t\\"Callback
```

Verify signature example

Java

Python

PHP

Golang

```
import javax.crypto.Mac;
import javax.crypto.spec.SecretKeySpec;
import java.util.Base64;
//# Function: Third-party callback sign verification
//# Parameters:
//#   key: The key configured in the console
//#   body: The body returned by the Tencent Cloud callback
//#   sign: The sign value returned by the Tencent Cloud callback
//# Return Value:
```

```
//#   Status: OK indicates that the verification has passed, FAIL indicates that th
//#   Info: Success/Failure information

public class checkSign {
    public static String getResultSign(String key, String body) throws Exception {
        Mac hmacSha256 = Mac.getInstance("HmacSHA256");
        SecretKeySpec secret_key = new SecretKeySpec(key.getBytes(), "HmacSHA256");
        hmacSha256.init(secret_key);
        return Base64.getEncoder().encodeToString(hmacSha256.doFinal(body.getBytes(
    }
    public static void main(String[] args) throws Exception {
        String key = "123654";
        String body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\
        String Sign = "kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA=";
        String resultSign = getResultSign(key, body);

        if (resultSign.equals(Sign)) {
            System.out.println("{'Status': 'OK', 'Info': 'validation passed'}");
        } else {
            System.out.println("{'Status': 'FAIL', 'Info': 'validation failed'}");
        }
    }
}
```

```
# -*- coding: utf8 -*-
import hmac
import base64
from hashlib import sha256

# Function: Third-party callback sign verification
# Parameters:
#   key: The key configured in the console
#   body: The body returned by the Tencent Cloud callback
#   sign: The sign value returned by the Tencent Cloud callback
# Return Value:
```

```
#    Status: OK indicates that the verification has passed, FAIL indicates that the
#    Info: Success/Failure information

def checkSign(key, body, sign):
    temp_dict = {}
    computSign = base64.b64encode(hmac.new(key.encode('utf-8'), body.encode('utf-8'
    print(computSign)
    if computSign == sign:
        temp_dict['Status'] = 'OK'
        temp_dict['Info'] = 'validation passed'
        return temp_dict
    else:
        temp_dict['Status'] = 'FAIL'
        temp_dict['Info'] = 'validation failed'
        return temp_dict

if __name__ == '__main__':
    key = '123654'
    body = "{\\n" + "\\t\\"EventGroupId\\":\\t2,\\n" + "\\t\\"EventType\\":\\t204,\
    sign = 'kkoFeO3Oh2ZHnjtg8tEAQhtXK16/KI05W3BQff8IvGA='
    result = checkSign(key, body, sign)
    print(result)
```



```
<?php

class TlsEventSig {

    private $key = false;
    private $body = false;

    public function __construct( $key, $body ) {
        $this->key = $key;
                $this->body = $body;
    }
```



```
    private function __hmacsha256() {
        $hash = hash_hmac( 'sha256', $this->body, $this->key, true );
                return base64_encode( $hash);
    }

    public function genEventSig() {
        return $this->__hmacsha256();
    }
}

$key="789";
$data="{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\\"Callbac

$api = new  TlsEventSig($key, $data);
echo $api->genEventSig();
```



```
package main
import "fmt"
import (
        "crypto/hmac"
        "crypto/sha256"
        "encoding/base64"
)

func main () {
    var data = "{\\n\\t\\"EventGroupId\\":\\t1,\\n\\t\\"EventType\\":\\t101,\\n\\t\
    var key = "789"
```

```
    //JSRUN engine 2.0, supporting up to 30 types of languages for online running,
    fmt.Println(hmacsha256(data,key))
}

func hmacsha256(data string, key string) string {
        h := hmac.New(sha256.New, []byte(key))
        h.Write([]byte(data))
        return base64.StdEncoding.EncodeToString(h.Sum(nil))
}
```

Access Management

Overview

Last updated：2023-10-08 16:02:33

notice

This document describes the management of access to **TRTC**. For access management of other Tencent Cloud services, see CAM-Enabled Products.

Cloud Access Management (**CAM**) is a web service provided by Tencent Cloud that helps customers securely manage access to their Tencent Cloud account resources. CAM allows you to create, manage, or terminate users or user groups and control who is allowed to access and use your Tencent Cloud resources through identity and policy management.

TRTC has supported **CAM**. You can grant TRTC permissions to sub-accounts as needed.

Getting Started

Before you start, make sure that you understand the basic concepts of CAM and TRTC, including:

CAM: User Types and Policy

TRTC: Application and SDKAppID

Use Cases

Granting product-level permissions

A company has multiple departments that are using Tencent Cloud's products. Department A is solely responsible for TRTC-related business, and the company needs to grant the department access to TRTC but not to other Tencent Cloud products. To achieve this, the company can create a sub-account for department A under its root account and grant the sub-account only TRTC-related permissions.

Granting application-level permissions

A company has multiple businesses that are using TRTC and needs to isolate them from each other. There are two dimensions to isolation: resource isolation and permission isolation. The former is enabled by TRTC's application system, and the latter by CAM. The company can create a sub-account for each of the businesses and grant them access only to the TRTC applications they are responsible for.

Granting action-level permissions

A company has a business that is using TRTC. It needs to grant the business' operational staff access to the TRTC console so that they can obtain usage statistics, and at the same time deny them access to critical operations such as modifying relayed push and on-cloud recording configurations. To achieve this, the company can create a custom policy that has the permissions to use relevant APIs to log in to the TRTC console and view usage statistics, and associate the policy with the sub-account created for the operational staff.

Authorization Granularity

In essence, CAM enables you to **allow or forbid specified accounts to access certain resources**. TRTC access management supports resource-level authorization. The granularity of manageable resources is TRTC applications, and the granularity of manageable actions is TencentCloud APIs, including server APIs and the APIs used to access the TRTC console. For more information, please see Manageable Resources and Actions.

Limitations

The granularity of manageable resources for TRTC access management is applications. Access control of finer granularity (e.g., application information or configuration information) is not supported.
TRTC does not support project-level access management. We recommend that you use tags to manage your cloud service resources.

Manageable Resources and Actions

Last updated：2023-10-08 16:03:14

notice

This document describes the management of access to **TRTC**. For access management of other Tencent Cloud services, see CAM-Enabled Products.

In essence, CAM enables you to **allow or forbid specified accounts to access certain resources**. TRTC access management supports resource-level authorization. The granularity of manageable resources is TRTC applications, and the granularity of authorizable actions is TencentCloud APIs, including server APIs and APIs that may be needed to access the TRTC console.

If you need to manage access to TRTC, please log in to the console with a Tencent Cloud root account and use a preset policy or a custom policy to grant permissions.

Type of Manageable Resources

TRTC access management allows you to control access to applications.

APIs Supporting Resource-Level Authorization

Barring a few exceptions, all API actions listed in this section support resource-level authorization. Authorization policies related to these API actions use the same **syntax conventions**. See below for details.

Authorizing access to all applications: `qcs::trtc::uin/${uin}:sdkappid/*`

Authorizing access to single applications: `qcs::trtc::uin/${uin}:sdkappid/${SdkAppId}` .

Server API actions

API	Category	Description
DismissRoom	Room management	Closes a room.
RemoveUser	Room management	Removes a user.
RemoveUserByStrRoomId	Room management	Removes a user (string room ID).
DismissRoomByStrRoomId	Room management	Closes a room (string room ID).
StartMCUMixTranscode	Stream mixing and transcoding	Starts On-Cloud MixTranscoding.

StopMCUMixTranscode	Stream mixing and transcoding	Stops On-Cloud MixTranscoding.
StartMCUMixTranscodeByStrRoomId	Stream mixing and transcoding	Starts On-Cloud MixTranscoding (string room ID).
StopMCUMixTranscodeByStrRoomId	Stream mixing and transcoding	Stops On-Cloud MixTranscoding (string room ID).
CreateTroubleInfo	Call quality monitoring	Generates information about exceptional conditions.
DescribeAbnormalEvent	Call quality monitoring	Queries abnormal events.
DescribeCallDetail	Call quality monitoring	Queries user list and call metrics.
DescribeHistoryScale	Call quality monitoring	Queries room and user numbers in the past.
DescribeRoomInformation	Call quality monitoring	Queries room list.
DescribeUserInformation	Call quality monitoring	Queries the list of historical users.

Console API actions

API	Console	Description
DescribeAppStatList	TRTC console: Overview Usage Statistics Monitoring Dashboard Development Assistance > UserSig Generation & Verification Application Management	Gets application list.
DescribeSdkAppInfo	TRTC console: Application Management > Application Info	Gets application information.
ModifyAppInfo	TRTC console: Application Management > Application Info	Modifies application information.
ChangeSecretKeyFlag	TRTC console: Application Management > Application	Enables/Disables encryption keys.

	Info	
CreateWatermark	TRTC console: Application Management > Material Management	Uploads an image.
DeleteWatermark	TRTC console: Application Management > Material Management	Deletes an image.
ModifyWatermark	TRTC console: Application Management > Material Management	Edits an image.
DescribeWatermark	TRTC console: Application Management > Material Management	Searches an image.
CreateSecret	TRTC console: Application Management > Quick Start	Generates a symmetric encryption key.
ToggleSecretVersion	TRTC console:Application Management > Quick Start	Switches between asymmetric keys (private and public keys) and symmetric keys.
DescribeSecret	TRTC console: Development Assistance > Demo Quick Run Development Assistance > UserSig Generation & Verification Application Management > Quick Start	Gets a symmetric encryption key.
DescribeTrtcAppAndAccountInfo	TRTC console: Development Assistance > UserSig Generation & Verification	Gets application and account information to obtain a pair of public and private keys.
CreateSecretUserSig	TRTC console: Development Assistance > UserSig Generation & Verification	Uses a symmetric encryption key to generate a UserSig.
DescribeSig	TRTC console:	Gets a UserSig generated using a pair of public and private keys.

	Development Assistance > UserSig Generation & Verification Application Management > Quick Start	
VerifySecretUserSig	TRTC console: Development Assistance > UserSig Generation & Verification	Verifies a UserSig generated using a symmetric encryption key.
VerifySig	TRTC console: Development Assistance > UserSig Generation & Verification	Verifies a UserSig generated using a pair of public and private keys.
CreateSpearConf	TRTC console: Application Management > Image Settings	Adds an image setting. This module is available only in iLiveSDK 1.9.6 and earlier versions. For TRTC SDK 6.0 and later versions, see Setting Image Quality
DeleteSpearConf	TRTC console: Application Management > Image Settings	Deletes an image setting. This module is available only in iLiveSDK 1.9.6 and earlier versions. For TRTC SDK 6.0 and later versions, see Setting Image Quality
ModifySpearConf	TRTC console: Application Management > Image Settings	Modifies image settings. This module is available only in iLiveSDK 1.9.6 and earlier versions. For TRTC SDK 6.0 and later versions, see Setting Image Quality
DescribeSpearConf	TRTC console: Application Management > Image Settings	Gets image settings. This module is available only in iLiveSDK 1.9.6 and earlier versions. For TRTC SDK 6.0 and later versions, see Setting Image Quality
ToggleSpearScheme	TRTC console: Application Management > Image Settings	Switches image setting scenarios. This module is available only in iLiveSDK 1.9.6 and earlier versions. For TRTC SDK 6.0 and later versions, see Setting Image Quality

APIs Not Supporting Resource-Level Authorization

Due to special restrictions, the following APIs do not support resource-level authorization.

Server API actions

API	Category	Description	Restriction
DescribeDetailEvent	Call quality monitoring	Queries specific events.	The parameters entered do not include `SDKAppID` , making resource-level authorization impossible.
DescribeRecordStatistic	Other APIs	Queries the billing period of on-cloud recording.	For business reasons, resource-level authorization is not supported currently.
DescribeTrtcInteractiveTime	Other APIs	Queries the billing period for audio/video interactive features.	For business reasons, resource-level authorization is not supported currently.
DescribeTrtcMcuTranscodeTime	Other APIs	Queries the billing period of relayed transcoding.	For business reasons, resource-level authorization is not supported currently.

Console API actions

API	Console	Description	Restriction
DescribeTrtcStatistic	TRTC console: Overview Usage Statistics	Gets usage statistics.	This API returns the statistics of all `SDKAppIDs`. Limiting a query to specific `SDKAppIDs` will lead to an error. You can use `DescribeAppStatList` to specify a list of applications to query.
DescribeDurationPackages	TRTC console: Overview Package Management	Gets the list of prepaid packages.	A prepaid package is shared by all TRTC applications under the same Tencent Cloud account. There is no `SDKAppID` parameter in the package information, so resource-level authorization cannot be performed.
GetUserList	TRTC console:	Gets user list.	The parameters entered do not include `SDKAppID`, making resource-level

	Monitoring Dashboard		authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
GetUserInfo	TRTC console: Monitoring Dashboard	Gets user information.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
GetCommState	TRTC console: Monitoring Dashboard	Gets call status.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
GetElasticSearchData	TRTC console: Monitoring Dashboard	Queries Elasticsearch data.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
CreateTrtcApp	TRTC console: Development Assistance > Demo Quick Run Application Management	Creates a TRTC application.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. `SDKAppID` is the unique ID of a TRTC application and is generated after application creation.
HardDescribeMixConf	TRTC console: Application Management > Function Configuration	Queries relayed push status.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
ModifyMixConf	TRTC console: Application Management > Function Configuration	Enables/Disables relayed push.	The parameters entered do not include `SDKAppID`, making resource-level authorization impossible. You can use `DescribeAppStatList` to specify a list of applications to query.
RemindBalance	TRTC	Gets the balance	A prepaid package is shared by all

	console: Package Management	alarm information of a prepaid package.	TRTC applications under the same Tencent Cloud account. There is no `SDKAppID` parameter in the package information, so resource-level authorization cannot be performed.

notice

You can use a custom policy to control access to an API that does not support resource-level authorization. In the policy statement, set the resource element to `*`.

Preset Policies

Last updated：2023-10-08 16:03:39

notice

This document describes the management of access to **TRTC**. For access management of other Tencent Cloud services, see CAM-Enabled Products.

TRTC access management works by associating permission policies with sub-accounts or granting policies to sub-accounts. The preset policies in the console allow you to perform some simple authorization. For more sophisticated authorization, see Custom Policies.

TRTC offers the following preset policies currently.

Policy	Description
QcloudTRTCFullAccess	Read-and-write permission
QcloudTRTCReadonlyAccess	Read-only permission

Examples of Using Preset Policies

Creating a sub-account with the read-and-write permission

1. Go to the User List page of the CAM console using a Tencent Cloud root account and click **Create User**.
2. On the displayed page, click **Custom Creation** to go to the "Create Sub-user" page.

explain

Finish the steps before **User Permissions** as instructed in Creating a Custom Sub-user.

3. On the **User Permissions** page:

3.1 Search for and check the preset policy `QcloudTRTCFullAccess` .

3.2 Click **Next**.

4. In the **Review** step, click **Complete**. After the sub-user is created successfully, download the login link and security credential file and store them properly. They contain the following information.

Information	Source	Use	Storage Required
Login link	Copied from the console page	Facilitates console login. Root account information is not required for login via the link.	No
User ID	Security credential file in CSV format	Required for console login	Yes
Password	Security credential file	Required for console login	Yes

	in CSV format		
SecretId	Security credential file in CSV format	Required for server API calling. For more information, seeAccess Key	Yes
SecretKey	Security credential file in CSV format	Required for server API calling. For more information, seeAccess Key	Yes

5. Provide the login link and security credentials to the party you want to authorize access, who will be able to use the sub-account to perform all kinds of TRTC operations, including visiting the TRTC console, calling TRTC server APIs, etc.

Granting read-and-write permission to existing sub-account

1. Go to the User List of the CAM console using a Tencent Cloud root account and click the target sub-account.
2. On the **User Details** page, click **Add** under the **Permission** tab. If the sub-account already has permissions, click **Associate Policy**.
3. Click **Select policies from the policy list**, search for and check the preset policy `QcloudTRTCFullAccess` , and complete the authorization as prompted.

Revoke the read-and-write permission of a sub-account

1. Go to the User List of the CAM console using a Tencent Cloud root account and click the target sub-account.
2. On the **User Details** page, find the preset policy `QcloudTRTCFullAccess` under the **Permission** tab, click **Disassociate** on the right, and complete the deauthorization as prompted.

Custom Policies

Last updated：2023-10-08 16:05:09

notice

This document describes the management of access to **TRTC**. For access management of other Tencent Cloud services, see CAM-Enabled Products.

It may be convenient to use a preset policy for access management in TRTC, but with preset policies, the granularity level of permissions is low, and permission granting cannot be specific to TRTC applications or TencentCloud APIs. To perform fine-grained authorization, you need to create custom policies.

Custom Policy Creation

There are multiple ways to create a custom policy. The table below offers a comparison of different methods. For detailed directions, see the remaining part of the document.

Access	Tool	Effect	Resource	Action	Flexibility	Complexity
CAM console	Policy generator	Manual selection	Syntax conventions	Manual selection	Medium	Medium
CAM console	Policy syntax	Syntax conventions	Syntax conventions	Syntax conventions	High	High
CAM server API	CreatePolicy	Syntax conventions	Syntax conventions	Syntax conventions	High	High

explain

TRTC does **not support** custom policy creation by product feature or project.

Manual selection means that you can select an object from a list of candidates offered in the console.

Syntax conventions means using the permission policy syntax to describe an object.

Permission Policy Syntax

Resource syntax conventions

The granularity level of manageable resources in TRTC access management is applications. Syntax conventions of permission policies for applications are in line with the Resource Description Method. In the example below, the developer (root account ID: `12345678`) has created three applications, whose `SDKAppIDs` are `1400000000` , `1400000001` , and `1400000002` .

Syntax convention of permission policy for all TRTC applications

```
"resource": [
  "qcs::trtc::uin/12345678:sdkappid/*"
]
```

Syntax convention of permission policy for single TRTC applications


```
"resource": [
  "qcs::trtc::uin/12345678:sdkappid/1400000001"
]
```

Syntax convention of permission policy for multiple TRTC applications

```
"resource": [
  "qcs::trtc::uin/12345678:sdkappid/1400000000",
  "qcs::trtc::uin/12345678:sdkappid/1400000001"
]
```

Action syntax conventions

The granularity level of authorizable actions in TRTC access management is TencentCloud APIs. For details, see Manageable Resources and Actions. The examples below use TencentCloud APIs such as `DescribeAppList` (gets application list) and `DescribeAppInfo` (gets application information).

Syntax convention of permission policy for all TencentCloud APIs

```
"action": [
  "name/trtc:*"
]
```

Syntax convention of permission policy for single TencentCloud APIs


```
"action": [
  "name/trtc:DescribeAppStatList"
]
```

Syntax convention of permission policy for multiple TencentCloud APIs

```
"action": [
  "name/trtc:DescribeAppStatList",
  "name/trtc:DescribeTrtcAppAndAccountInfo"
]
```

Examples of Using Custom Policies

Using the policy generator

In the example below, we create a custom policy that allows all actions under TRTC application `1400000001` except calling the server API `RemoveUser` .

1. Go to the **Policy** page of the CAM console using a Tencent Cloud root account and click **Create Custom Policy**.
2. Select **Create by Policy Generator**.
3. Select the service and action.

For **Effect**, select **Allow**.

For **Service**, select **Tencent Real-Time Communication (trtc)** .

For **Action**, check all the items.

For **Resource**, enter `qcs::trtc::uin/12345678:sdkappid/1400000001` , which aligns with the syntax described in Resource syntax conventions.

No configuration is needed for **Condition**.

Click **Add Statement**, and a statement indicating that any action is allowed under TRTC application `1400000001` appears below.

4. Add another statement on the same page.

For **Effect**, select **Deny**.

For **Service**, select **Tencent Real-Time Communication (trtc)**.

For **Action**, select `RemoveUser` . You can use the search feature to quickly locate the action.

For **Resource**, enter `qcs::trtc::uin/12345678:sdkappid/1400000001` , which aligns with the syntax described in Resource syntax conventions.

No configuration is needed for **Condition**.

Click **Add Statement**, and a statement indicating that calling `RemoveUser` is forbidden under TRTC application `1400000001` appears below.

5. Click **Next** and rename the policy if necessary.
6. Click **Done** to complete the creation.

You can then grant the policy to other sub-accounts as described in Granting read-and-write permission to existing sub-account.

Using the policy syntax

In the example below, we create a custom policy that allows all actions under TRTC application `1400000002` and all actions but calling `RemoveUser` under `1400000001` .

1. Go to the Policy page of the CAM console using a Tencent Cloud root account and click **Create Custom Policy**.
2. Select **Create by Policy Syntax**.
3. In the **Select a template type** section, select **Blank Template**.

explain

A policy template allows you to create a policy by modifying a copy of an existing policy (preset or custom). You can choose a policy template that fits your actual conditions to reduce the complexity and workload of writing permission policies.

4. Click **Next** and rename the policy if necessary.
5. Enter the following content in the **Policy Content** box.

```
{
 "version": "2.0",
 "statement":[
     {
         "effect": "allow",
         "action": [
             "name/trtc:*"
         ],
```

```
            "resource": [
                "qcs::trtc::uin/12345678:sdkappid/1400000001",
                "qcs::trtc::uin/12345678:sdkappid/1400000002"
            ]
        },
        {
            "effect": "deny",
            "action": [
                "name/trtc:RemoveUser"
            ],
            "resource": [
                "qcs::trtc::uin/12345678:sdkappid/1400000001"
            ]
        }
    ]
  }
```

explain

Policy content must align with the Syntax Logic. About the syntax of the resource and action elements, see Resource syntax conventions and Action syntax conventions above.

6. Click **Create Policy** to complete the creation.

You can then grant the policy to other sub-accounts as described in Granting read-and-write permission to existing sub-account.

Using server APIs provided by CAM

Managing access in the console can meet the business needs of most developers, but to automate and systematize your access management, you need to use server APIs.

Permission policy-related server APIs belong to CAM. For details, see CAM documentation. Only a few main APIs are listed below:

CreatePolicy

DeletePolicy

AttachUserPolicy

DetachUserPolicy

How to push stream to TRTC room with OBS WHIP

Last updated：2024-08-07 10:53:53

Overview

OBS includes WHIP support, which allows you to do many interesting things by combining the powers of both OBS and WHIP.

WHIP is a standard protocol that lets you use HTML5 and different clients to publish and play live streams. Plus, you can use open-source tools to build your own live streaming platform.

You can also use TRTC (Tencent Real-Time Communication) cloud services with OBS WHIP support for a streaming platform. This is a great option if you don't want to build your own platform or need a more reliable and stable platform with dedicated support.

Additionally, TRTC (Tencent Real-Time Communication) provides a free trial that includes a specific amount of streaming time, making it super easy for you to try out.

If you need help or run into any problems, don't hesitate to contact us on Discord.

Prerequisites

Before you move forward, double-check that you've got these necessary items ready:

OBS with WHIP support, please download from OBS

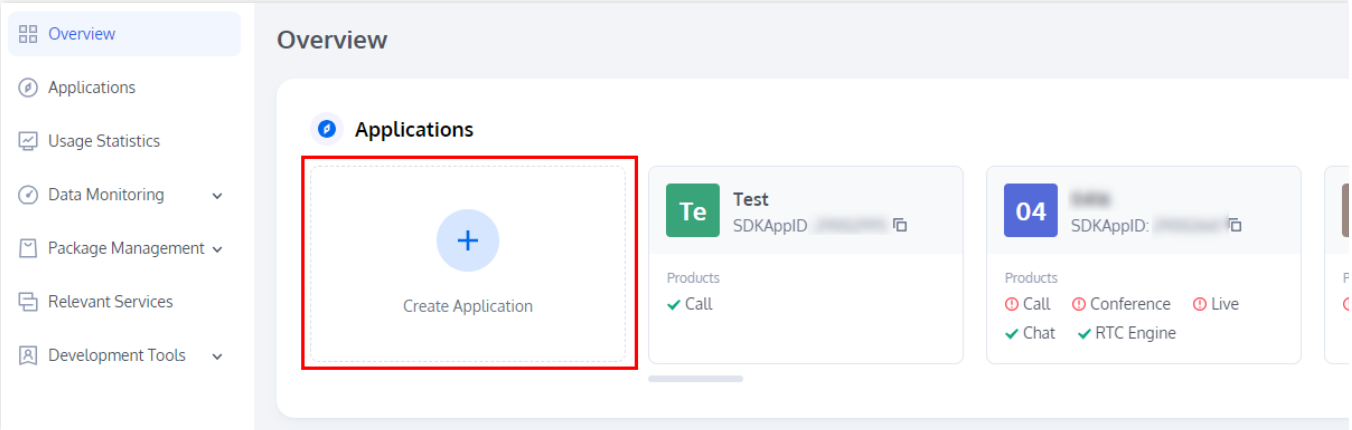
TRTC (Tencent Real-Time Communication) account, please register here

Next, you need to create a TRTC application and generate a Bearer Token for WHIP.

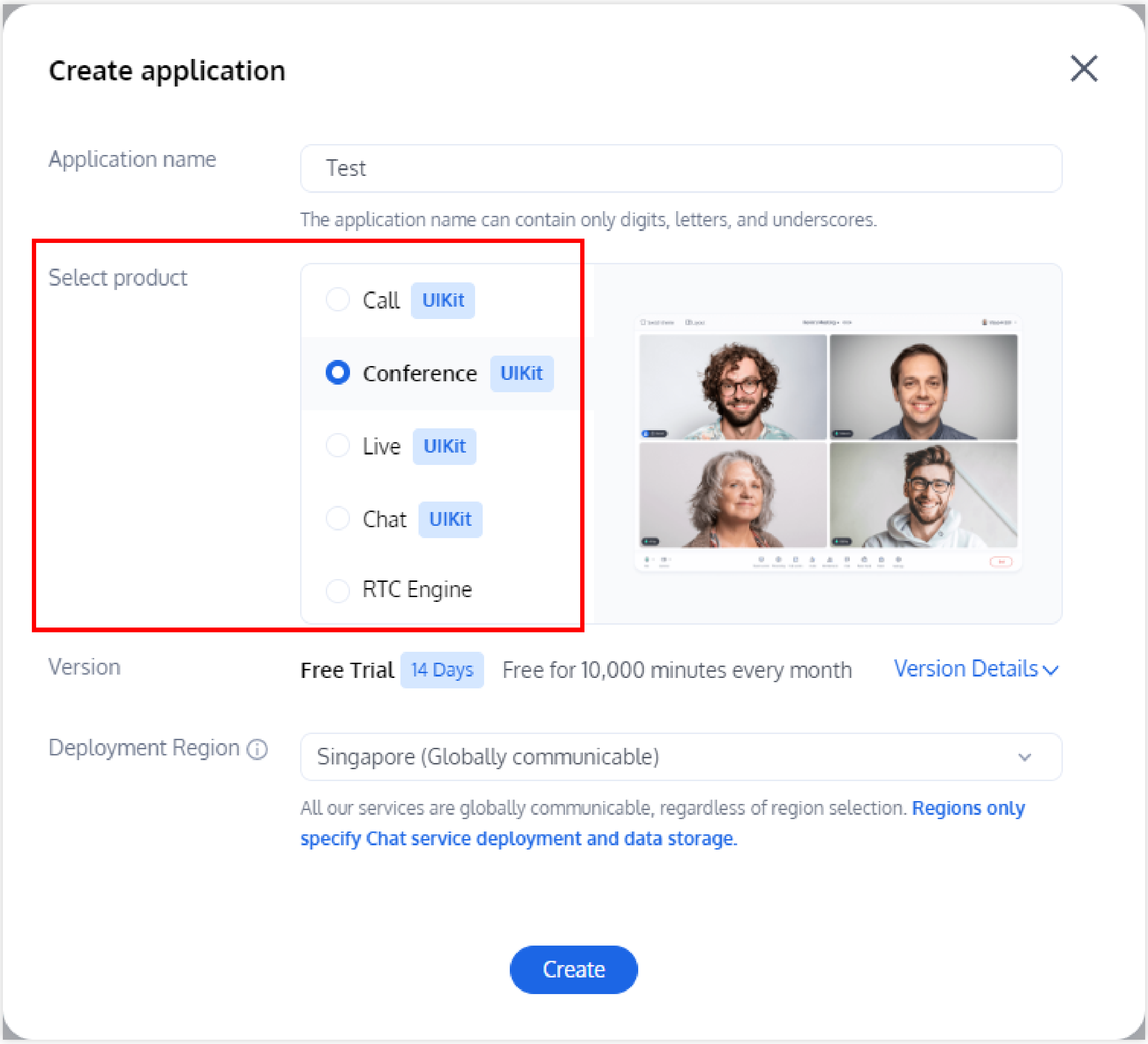
Step 1: Create a TRTC application

Please follow the steps below to create a TRTC application:

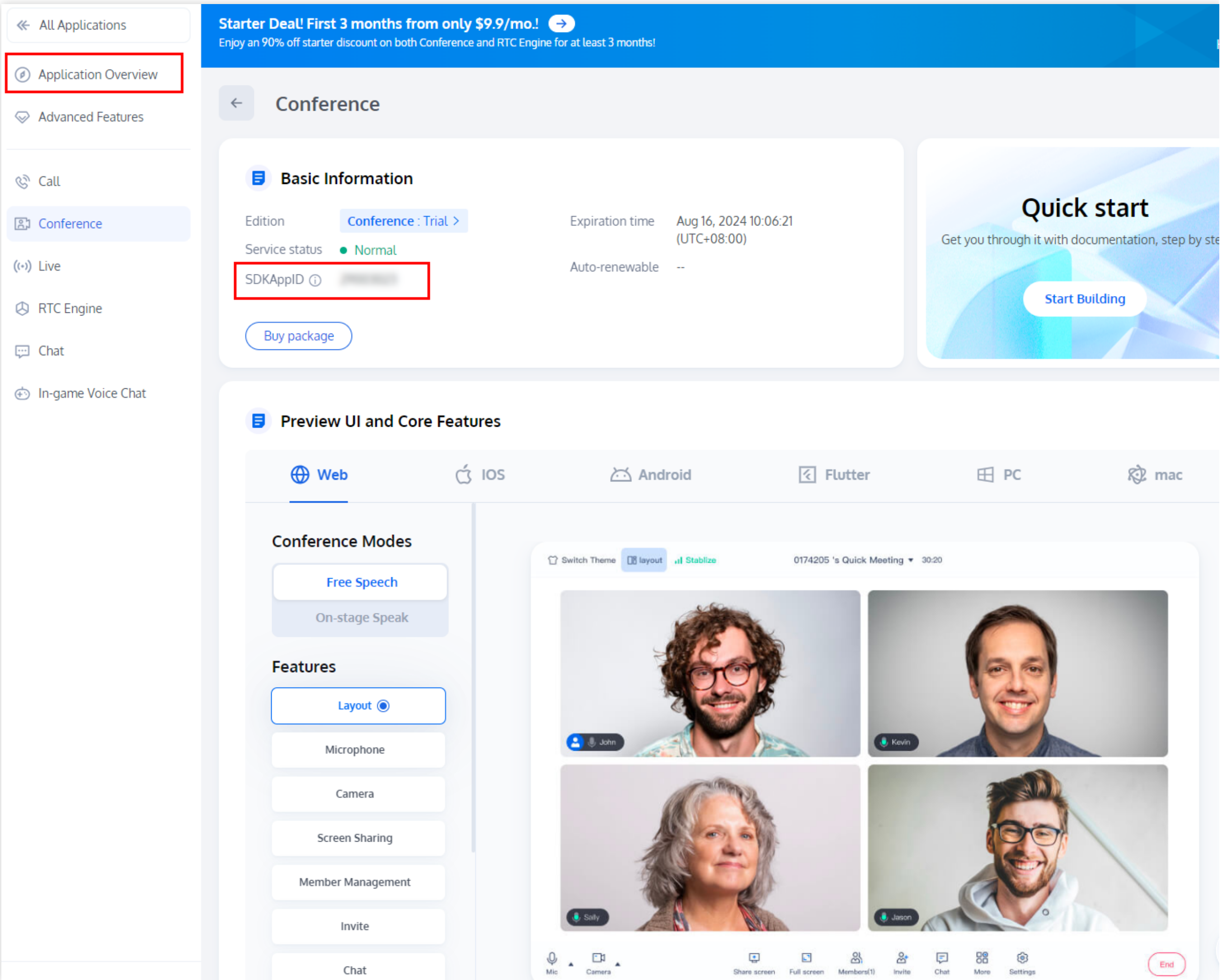
1. Log in to the Tencent RTC Console and click **Create Application**.



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**.



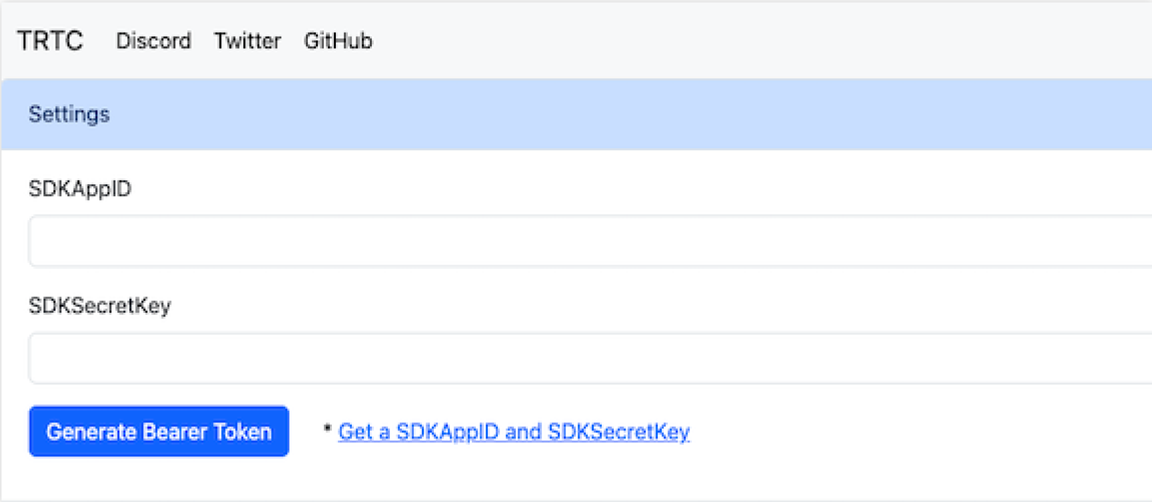
3. After completing the application creation, you will automatically enter the application details page of the selected product. You can view the `SDKAppID` and `SDKSecretKey` in the **Application Overview**, which will be used in subsequent steps.



Step 2: Create a Bearer Token for WHIP

Following that, you must generate a Bearer Token for WHIP, which will be utilized in OBS.

You can directly visit https://tencent-rtc.github.io/obs-trtc/bearer.html to create a WHIP Bearer Token. Ensure that use the AppID with your own `SDKAppID` and secret with your own `SDKSecretKey` , then click the `Generate Bearer Token` button.



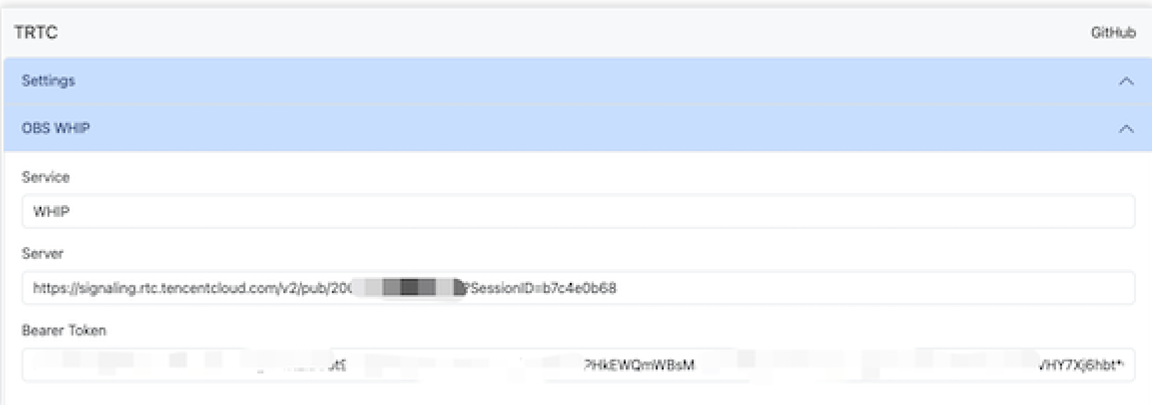
Note：

You can also access the url `https://tencent-rtc.github.io/obs-trtc/bearer.html?appid=2000xxx&secret=yyyyyy` to setup the parameters.

Next, use the generated WHIP Bearer Token to configure OBS.

Step 3: Configure OBS

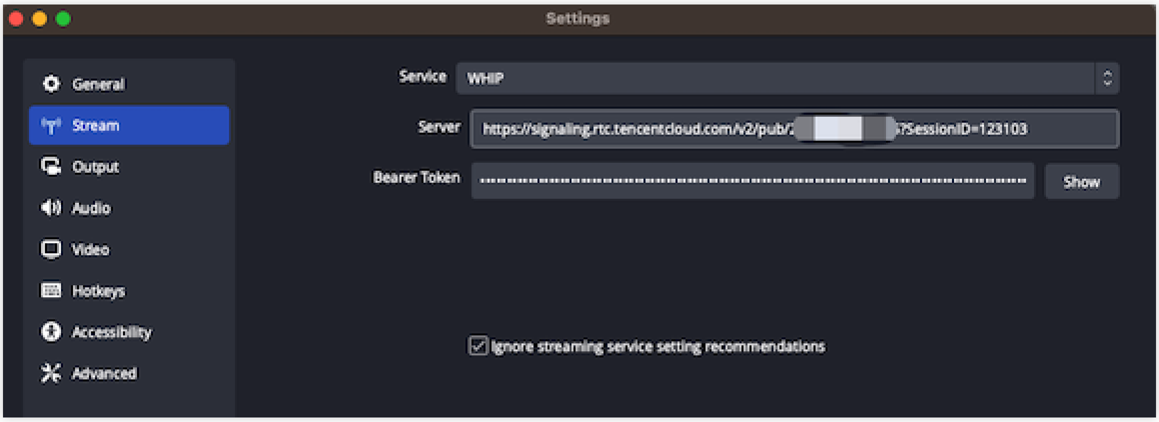
In the `OBS WHIP` section, you will find the generated WHIP `Server` and `Bearer Token` for configuring OBS.



Please follow the steps below to configure OBS:

1. Open OBS and click **Settings**.
2. Click **Stream** on the left sidebar.
3. Select `WHIP` for **Service**.
4. Make sure to input the `Server` and `Bearer Token` accurately.

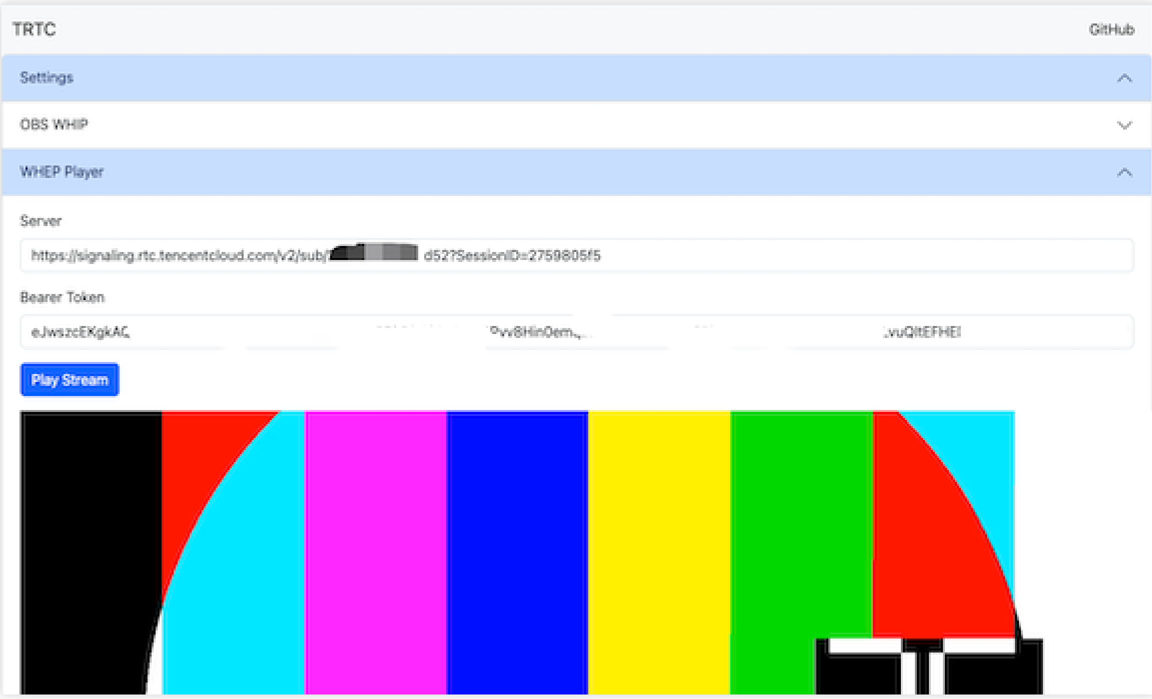
5. Click **OK** to save the settings.
6. Click **Start Streaming** to start.



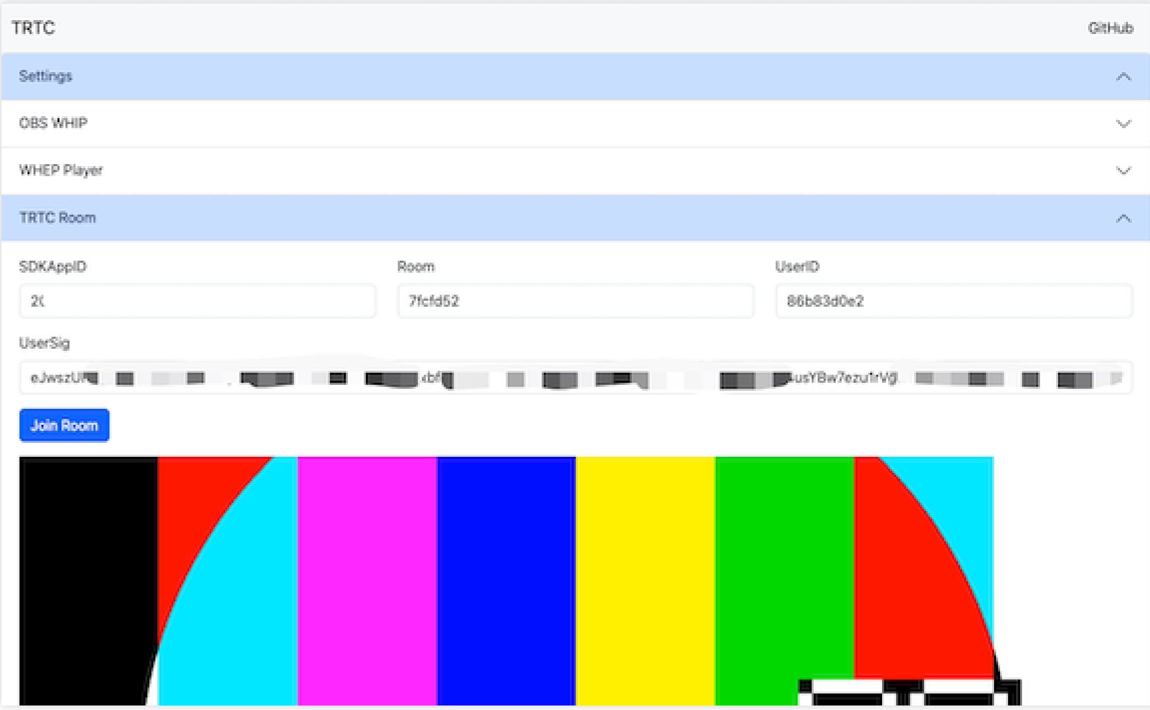
At this point, the stream is streaming to the TRTC service.

Step 4: Play the stream

Open the previous webpage, go to the `WHEP Player` section, and click **Play Stream** to play the stream via WHEP.



Another option is to go to the `TRTC Room` section, and click **Join Room** to access the TRTC room and watch the stream via TRTC, or you can utilize the TRTC mobile SDK to join the room and view the stream.



Since both WHIP and WHEP are standard protocols, you can utilize any client that supports them to play the stream.

Conclusion

We looked into using TRTC (Tencent Real-Time Communication) cloud services to make a stronger streaming platform and the steps needed to create a TRTC app with OBS WHIP help. These tools make it easier to organize and provide real-time live streaming experiences for different situations, with the power of OBS.

If you require assistance or encounter any difficulties, please feel free to reach out to us via Discord.

