SparkRTC

FAQs

Issue 01

Date 2026-05-13

HUAWEI CLOUD COMPUTING TECHNOLOGIES CO., LTD.

Trademarks and Permissions


and other Huawei trademarks are the property of Huawei Technologies Co., Ltd.
All other trademarks and trade names mentioned in this document are the property of their respective holders.

Notice

The purchased products, services and features are stipulated by the contract made between Huawei Cloud and the customer. All or part of the products, services and features described in this document may not be within the purchase scope or the usage scope. Unless otherwise specified in the contract, all statements, information, and recommendations in this document are provided "AS IS" without warranties, guarantees or representations of any kind, either express or implied.

The information in this document is subject to change without notice. Every effort has been made in the preparation of this document to ensure accuracy of the contents, but all statements, information, and recommendations in this document do not constitute a warranty of any kind, express or implied.

Huawei Cloud Computing Technologies Co., Ltd.

Address: Huawei Cloud Data Center Jiaoxinggong Road
Qianzhong Avenue
Gui'an New District
Gui Zhou 550029
People's Republic of China

Website: https://www.huaweicloud.com/intl/en-us/

Contents

1 About SparkRTC

1.1 What Is SparkRTC?

SparkRTC, built on Huawei's long-established video technologies and expertise, delivers a high-concurrency and high-definition experience in real time. It offers dependable security at low latency and is an ideal option for a broad array of real-time communication (RTC) scenarios, such as online education, cloud conferencing, and social entertainment. For details, see **SparkRTC Service Overview**.

1.2 What Are the Differences Between Real-Time Audio and Video and Livestreaming?

The livestreaming latency can be 3s to 5s. The streaming end pushes collected live streams to the Live center to process and distribute the streams to playback devices. The streaming protocol can be Real-Time Messaging Protocol (RTMP) and the playback protocol can be RTMP, HTTP-FLV, or HTTP Live Streaming (HLS).

The real-time audio and video, with a latency lower than 400 ms, is an ideal option for one-to-one or one-to-many audio and video calls, video co-hosting, and low-latency livestreaming. Huawei Cloud SparkRTC provides SDKs that are compatible with all mainstream platforms.

1.3 Can SparkRTC Be Used Outside China?

Yes. SparkRTC provides a high-quality large real-time communication (RTC) network across the globe.

1.4 What Is SparkRTC RoomID and How Can I Name One?

RoomID indicates the ID of a room. A room is an audio and video space where users can receive real-time audio and video streams from each other.

- SparkRTC isolates users in different virtual rooms.
- Only users in the same room can receive audio and video streams from each other.
- A **RoomID** can be customized and contain up to 64 characters, including letters, numbers, underscores (_), and hyphens (-).
- The room ID is defined by the tenant.

1.5 How Many Rooms Can Be Created in SparkRTC at Most at a Time?

There is no limit on the number of online rooms in SparkRTC.

1.6 What Is SparkRTC UserID and How Can I Name One?

SparkRTC uses user IDs to identify users of an application. Each user has a unique user ID.

- Developers can customize user IDs based on specifications.
- A **UserID** can be customized and contain up to 64 characters, including letters, numbers, underscores (_), and hyphens (-).

1.7 What Are the Roles of SparkRTC?

Users in a room have different roles and consequently different permission models. There are three types of roles:

- **Publisher**: streamer who sends streams but does not receive streams. It is a role type reserved by SparkRTC.
- **Joiner**: interactive viewer who sends and receives streams
- **Player**: common viewer who only receives streams

In the SparkRTC demo, the role switch and co-hosting are performed by **Joiners** and **Players**.

1.8 How Many Participants Are Allowed by SparkRTC in a Video Call?

A single room can contain a maximum of 500 publishers at a time and there is no limitation on the number of viewers.

1.9 What Is the Latency of SparkRTC?

The abundant node resources of SparkRTC ensure an average end-to-end (E2E) latency lower than 200 ms.

1.10 Does SparkRTC Support Co-Hosting?

SparkRTC supports the co-hosting of users in the same room.

1.11 Does SparkRTC Support Bidirectional Desktop Stream Push?

No. The screen, window, or area can be shared by only one person in a room at a time.

1.12 Does SparkRTC Support Private Deployment?

Huawei Cloud SparkRTC does not support private deployment. To enable this function, contact Huawei technical support by referring to **submit a service ticket**.

1.13 Does SparkRTC Support USB Cameras on PCs?

Yes. You can connect a USB camera to the PC to use SparkRTC.

2 Billing

2.1 When I Subscribe to SparkRTC Video and Audio Services, Is the Video Charged by Resolution Plus the Audio Charge?

When you enter a room and receive video streams, only the video duration is counted and you will not be charged for the audio. The audio duration is counted only when you do not subscribe to the video service.

2.2 Does the Billing Vary with Bitrates When There Are Only Audios?

The billing does not vary with bitrates when there are only audios.

2.3 How Do I View Bills and Fee Deduction Details?

You can choose **Billing** > **Bills** in **Billing Center** to view bills and fee deduction details.

2.4 Can I Use the Postpaid Billing Mode?

Huawei Cloud SparkRTC is pay-per-use by default.

2.5 Why Am I Charged Even When There Is Only One Person in the Room and There Is No Audio/video Call?

SparkRTC billing is based on how long each user of a room watches the video streams of other uses. The billing can also vary with resolution levels. If there is only one person in a room, SparkRTC cloud resources are used even if no stream is pushed (no uplink data is generated). When there is only one person in a room,

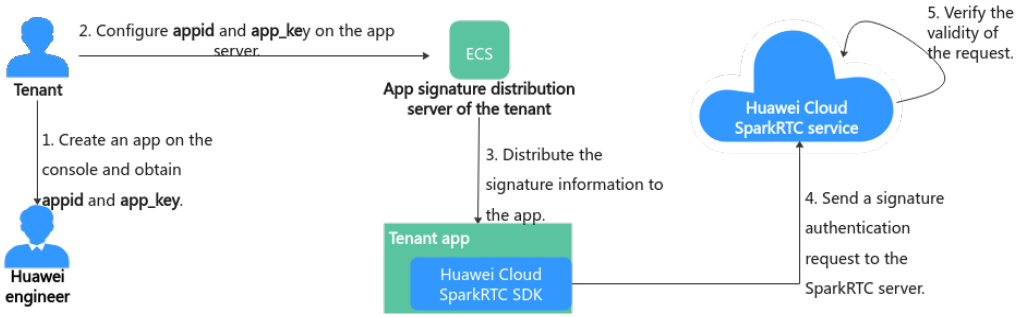
the person cannot subscribe to the audio/video streams of others and cannot see the video image. The service usage is calculated based on the voice duration.

3 Signature

3.1 What Is Signature?

To ensure the communication security of SparkRTC, access authentication is required when a user joins a channel.

Figure 3-1 Authentication principles



Huawei Cloud SparkRTC uses digital signatures for access authentication. Information such as **app_id**, **room_id**, **user_id**, **ctime**, and **signature** must be provided in the initialization or login function of the SDK to access SparkRTC. **signature** indicates the signature, which is generated by the tenant using the **app_key** provided by SparkRTC based on the Huawei SparkRTC signature algorithm. App developers need to obtain authentication signatures from remote servers. For details, see **Access Authentication**.

3.2 How Do I Obtain a Key?

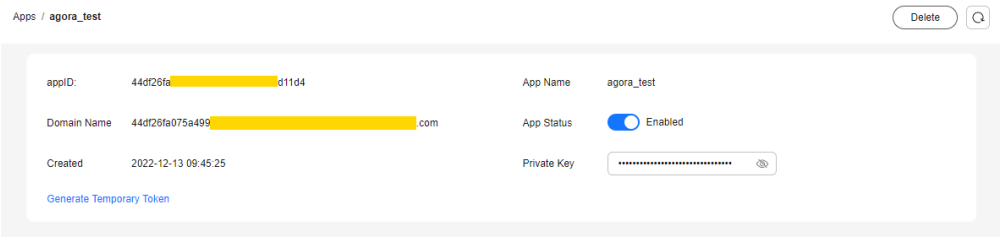
app_key is the authentication key generated by SparkRTC for each app. The key must be securely stored. You can obtain the authentication key by performing the following operations:

1. Log in to the **SparkRTC console**.
2. In the navigation pane on the left, choose **Apps**.

3. On the displayed page, click an app name in the **Name/ID** row to view its details.

   Copy the value of **Private Key** if it is visible.

   Figure 3-2 Application details page



3.3 How Does the Server Calculate the Value of signature?

```
// Hard-coded or plaintext app_key and app_id in code pose significant security risks. It is highly recommended that these credentials be stored in ciphertext within files or environment variables, and decrypted as needed to ensure security. In this example, app_key and app_id are stored in environment variables. Before running the code, set the environment variables APP_KEY and APP_ID in your local environment.
app_key =  System.getenv("APP_KEY");
app_id =  System.getenv("APP_ID");
signature = HMAC256(app_key,(app_id + room_id + user_id +ctime))
```


4 Recording

4.1 What Are the Differences Between Single Stream Recording and Mixed Stream Recording?

The differences between single stream recording and mixed stream recording are as follows:

- Single stream recording: The audio/video streams of each user in a room can be recorded and saved in independent files. For example, if parents only want to watch videos of their child and the teacher, they can adopt single stream recording. Videos of their child and the teacher are recorded separately and then combined. For details about how to implement single stream recording, see **Creating a Single Stream Recording Job**.
- Mixed stream recording: Audio/Video streams in a room can be mixed in the cloud, and then the mixed audio/video streams can be recorded and saved in one file. For example, if you want to record a live video with co-hosting, you can select mixed stream recording to record the audio/video streams of all hosts and store them in one file. You do not need to combine the recorded audio/video streams using script. For details about how to implement mixed stream recording, see **Creating a Mixed Stream Recording Job**.

4.2 How Do I Restore a Deleted Recording File?

Recording files are stored in OBS buckets. To restore a deleted recording file, ensure that versioning has been enabled for an OBS bucket before deleting the recording file.

- If versioning is enabled for a bucket, deleted objects are saved to the **Deleted Objects** list. You can recover objects from the **Deleted Objects** list.
- If versioning is disabled for a bucket, deleted objects cannot be restored.

4.3 How Do I Authenticate a Recording Callback?

To ensure recording callback communication security, tenants can configure the recording callback parameter **Private Key** to authenticate recording callback

messages. This section describes how to generate a recording callback authentication signature.

Parameter Description

Table 4-1 Parameter description

Parameter	Description
X-Rtc-Rand	Random number in the request header.
X-Rtc-Timestamp	Timestamp in the request header.
X-Rtc-Signature	Signature in the request header.
msg	Content of the request body.

Method of Generating a Recording Callback Authentication Signature

1. Combine the following parameters into a string.

   The code is as follows:

   ```
   String content = X-Rtc-Rand + X-Rtc-Timestamp + msg;
   ```

2. Use the callback key (**key**) in the recording callback configuration to encrypt the combined string **content** in HMAC-SHA256 mode to obtain the signed string.

   The code is as follows:

   ```
   String key = System.getenv("KEY");
   String signatureStr = hmacSha(key, content, "HmacSHA256");
   static String hmacSha(String KEY, String VALUE, String SHA_TYPE) {
       try {
           SecretKeySpec signingKey = new SecretKeySpec(KEY.getBytes("UTF-8"), SHA_TYPE);
           Mac mac = Mac.getInstance(SHA_TYPE);
           mac.init(signingKey);
           byte[] rawHmac = mac.doFinal(VALUE.getBytes("UTF-8"));

           byte[] hexArray = {
                   (byte) '0', (byte) '1', (byte) '2', (byte) '3',
                   (byte) '4', (byte) '5', (byte) '6', (byte) '7',
                   (byte) '8', (byte) '9', (byte) 'a', (byte) 'b',
                   (byte) 'c', (byte) 'd', (byte) 'e', (byte) 'f'
           };
           byte[] hexChars = new byte[rawHmac.length * 2];
           for (int j = 0; j < rawHmac.length; j++) {
               int v = rawHmac[j] & 0xFF;
               hexChars[j * 2] = hexArray[v >>> 4];
               hexChars[j * 2 + 1] = hexArray[v & 0x0F];
           }
           return new String(hexChars);
       } catch (Exception ex) {
           throw new RuntimeException(ex);
       }
   }
   ```


5 SDK Usage

5.1 How Do I Create a Room?

The room is automatically created by the SparkRTC background when you enter the room. You only need to call client APIs to join the room. For details about the APIs, see **SparkRTC API Reference**.

5.2 Why Can't the SparkRTC Apps Running on Two Mobile Phones Communicate with Each Other?

Check whether the *roomId*s of the two mobile phones are the same. Interaction in one room is supported only when the *roomId*s are the same and the *userId*s are different. SparkRTC does not allow using one *userId* on two devices at the same time.

5.3 How and When Is Authentication Signature Generated?

An authentication signature is generated when a user joins a room and is passed to the parameters for joining the room. For details about how to generate an authentication signature, see **Access Authentication**.

5.4 Why Does the Program Crash when the onVideoStats, onAudioStatus, or onSubStreamStats Callback Is Triggered?

The pointers of the input parameters **localStats** and **remoteStats** of the callback function may be null. Ensure that the pointers are not null before you use the parameters. Otherwise, a null pointer error may occur.

5.5 Why Can I Hear My Own Voice from the Local Receiver?

This is because you set the user ID to your own when you called the **muteRemoteAudio** API.

5.6 Can I Call setExternalAudioCapture, setExternalVideoCapture, or setExternalMediaFrameOutput After Entering a Room?

No. You need to call these APIs before joining a room and cannot change them after joining the room.

5.7 Does SparkRTC Support Video Mirroring?

Yes. You can call the **updateLocalRenderMode** API to configure the mirroring mode of the preview images captured by the local camera.

5.8 How Do I Obtain Data Such as the Uplink and Downlink Bitrates, Resolution, Packet Loss Rate, and Audio Sampling Rate During an Audio/Video Call?

You can call the **onRtcStats()** API to obtain data such as the uplink and downlink bitrates, resolution, packet loss rate, and audio sampling rate.

5.9 Can I Set the Call Volume of a Local User? Can I Set the Playback Volume of Each Remote User?

Yes. Call **adjustRecordingVolume()** to set the call volume of a local user, and **adjustPlaybackVolume()** to set the playback volume of a remote user.

5.10 What Is the Difference Between enableLocalVideo and muteLocalVideo?

The difference is as follows:

- **enableLocalVideo**: indicates whether to enable local video data collection. When this function is disabled, black screens are displayed on the local and remote videos.
- **muteLocalVideo**: indicates whether to send the local video to the background. When this function is disabled, black screens are displayed on the

remote video of other users, but the local video can still be viewed in the preview mode.

5.11 What Is the Difference Between enableLocalAudioStream and muteLocalAudio?

The difference is as follows:

- **enableLocalAudioStream**: indicates whether to enable local audio data collection. The function is enabled by default. When it is disabled, local audio data collection and upstream transmission are disabled.
- **muteLocalAudio**: indicates whether to send audio streams to the background. When this function is disabled, audio and video data will continue to be sent in the form of mute packets at an extremely low bitrate.

5.12 Does SparkRTC Support Permission Verification for Rooms?

User authentication is supported. Huawei Cloud SparkRTC uses digital signatures for access authentication. Information such as **app_id**, **room_id**, **user_id**, **role**, **ctime**, and **signature** must be provided in the initialization or login function of the SDK to access SparkRTC. For details, see **Access Authentication**.

5.13 Does the PC Support Screen Sharing?

Yes. For details, see **Screen Sharing (Web)**.

5.14 Does SparkRTC Allow Others Users to View a Local Video During Real-Time Communications?

Yes. This function can be implemented in either of the following ways:

- Share the desktop so that other users in the room can view the local video.
- Customize data collection of the local video file so that other users in the room can view the video.

For details, see **API Reference**.

5.15 Can I View Only Real-Time Audio Streams on SparkRTC?

Yes. Users at the peer end can choose not to view the video.

5.16 Can More Than One Presentation Stream Be Shared in a Room?

No. Only one presentation stream can be shared in a room.

5.17 Does SparkRTC Support 1080p?

The maximum resolution supported by SparkRTC varies depending on the stream type:

- Video streams: streams collected by cameras. SparkRTC can encode, receive, and send video streams at four resolution levels (720p, 360p, 180p, and 90p) of one video source at the same time.
- Presentation streams: streams of the shared screen or window. The default resolution is 1080p. If the fluency is preferred, the presentation resolution will be changed to 720p.

5.18 Is Video Snapshot Capturing Supported?

No. You can use the snapshot capturing API provided by the PC, Android, or iOS.

5.19 Can I Play Online Music for Background Music?

No.

5.20 Can I Create *N* RTC Objects on the Same Page and Log In to *N* Rooms Using *N* User IDs?

Yes.

5.21 Is the Streaming Resolution (Width and Height) on the Web Client Applicable to All Browsers?

Due to the restrictions of the device and browser, the video resolution may not be completely matched. In this case, the browser automatically adjusts the resolution so that it is close to the resolution corresponding to the profile. For details, see **setVideoProfile**.

5.22 How Do I Clear Data in the Camera List After a Camera Is Removed from the SDK on the Web Client?

You can call **getCameras** to check whether the new device list can be obtained. If the information about the removed camera can still be obtained, the browser does not refresh the list, and the web SDK cannot obtain the new device list.

5.23 Can the SDK on the Web Client Obtain the Current Volume?

You can call **getAudioLevel** to obtain the current volume.

5.24 Can I Receive Notifications on a Web Client When a Remote User Leaves a Room?

Yes. You are advised to use the client.on('**peer-leave**') event to receive notifications when remote users leave a room.

5.25 Why Is There Echo, Undesired Sound, Noise, or Low Volume During a Call on the Web Client?

This can occur when devices of the two parties are too close to each other and is normal. Keep the devices far away from each other during the test. If echo, undesired sound, or noise is heard from the web client, 3A processing does not take effect on the web client.

If you use the native **getUserMedia** API of the browser for customized stream collection, you need to manually configure 3A parameters. For details, see **MediaTrackConstraints**.

- **echoCancellation**: echo cancellation
- **noiseSuppression**: noise suppression
- **autoGainControl**: automatic gain control

If you use the **HRTC.createStream** API to collect streams, you do not need to manually configure 3A parameters. 3A is enabled by default in the SDK.

5.26 How Do I Switch to Another Camera or Microphone on the Web Client?

Audio/video input devices are identified by device ID (**deviceId**). Each audio/video device has a unique device ID. You can call **getDevices** to obtain the device ID. The device ID is randomly generated. In some cases, the ID of the same device may change. You are advised to call **getDevices** to obtain the device ID each time you

switch to another device. You can call **getDevices** to obtain the camera and microphone, and then call **switchDevice** to switch to another device.

5.27 Does SparkRTC Support the Whiteboard Function?

No. The whiteboard function is an app-level application.

5.28 Can I Enter Multiple Rooms with the Same User ID?

Yes.

5.29 How Long Is the Lifecycle of a Room on SparkRTC?

The following situations may occur:

- The first user who joins a room is the owner of the room, but the user cannot dismiss the room.
- When all users leave a room, the background immediately dismisses the room.
- If one user in a room is disconnected, the user will be removed from the room after the keepalive time expires. If all users in a room are disconnected, the room will be automatically dismissed after the keepalive time expires.
- If the room that a user wants to join does not exist, the background automatically creates a room.

5.30 Can I Use Bluetooth Headsets for Video Calls on SparkRTC?

Yes.

5.31 Can I Subscribe to Only Audio of 2/3/4 Participants During a Video Call?

Yes. The top-N-audio mode of SparkRTC is also called the top three loudest participant mode. After the top-N-audio mode is enabled, a local user can receive the audio of the three loudest users in the room without calling an API to subscribe to the audio of a remote user. For details about how to call the API, see **Switching the Audio Mode**.

5.32 Will the Resolution of the Video Stream Change When the Size of a Specified Shared Window Changes?

Yes.

5.33 Can I Switch Between the Floating Window and Large and Small Windows During a Video Call?

This function is part of the UI layout logic, which is not restricted by the SDK.

5.34 Can I Use the API for Sending Customized Messages to Implement Functions Such as Chat Rooms and On-Screen Comments?

No.

5.35 Can I Query All User IDs?

This function is not supported currently.

5.36 Why Can't I Call the checkSystemRequirements API in the Web SDK?

Check whether the Web SDK version is between 2.0.2 and 2.0.9.300. These versions support the **strictCheckBrowser** parameter, which is used to strongly verify the WeChat browser whitelist function and has compatibility risks.

Ensure that the version is 2.0.9.301 or later before October 29, 2023. Users of WeChat browsers need to update the SDK version.

6 Troubleshooting

6.1 What Can Enterprise Customers Do when Room Join Failure, Silence or Black Screen Occurs During SparkRTC Calls?

Enterprise users usually have network security settings, which may cause SparkRTC call failures.

Check the internal network first to ensure that the customer's network can communicate with the public network. Alternatively, the firewall must meet the following requirements:

1. Access restrictions for all IP addresses.
2. Configure the SparkRTC port and domain name whitelists for the client firewall.
   - Configure the SparkRTC port whitelist for the firewall, as shown in **Table 6-1**.

     Table 6-1 Port whitelist for the firewall

Port Type	Port
TCP	443, 6447
UDP	20000–20063

   - Configure the SparkRTC domain name whitelist for the firewall, as shown in **Table 6-2**.

     Table 6-2 Domain name whitelist for the firewall

No.	Domain Name
1	*.dbankcdn.com

No.	Domain Name
2	*.dbankcdn.cn
3	*.dbankcloud.ru
4	*.dbankcloud.cn
5	*.dbankcloud.com
6	*.hicloud.cn
7	*.hicloud.com
8	*.dbankedge.cn

If the problem persists, you are advised to **submit a service ticket** for Huawei Cloud technical support.

6.2 Why Can't I Hear When Joining a Meeting?

The possible causes are as follows:

- You did not subscribe to the meeting before joining the meeting.
- The server is faulty.
- Remote video streams are not enabled.

6.3 Why Does Video Freezing Occur?

Video freezing may occur due to poor network or device performance. You can perform the following operations to locate the fault:

- You can check:
  - Whether the freezing is continuous or one-time. One-time freezing is normal and occurs occasionally.
  - Whether the network connection is normal.
  - If the network connection is normal but the problem persists, change the network connection.
  - If the network connection is good, use another device to check whether the problem persists.
  - If video pre-processing such as beautification is required, disable it and check whether the freezing is caused by pre-processing.

If the problem persists, you are advised to **submit a service ticket** for Huawei Cloud technical support.

6.4 What Do I Do If a Black Screen Occurs During Video Playback?

A black screen may occur on:

- Local video
- Remote video
- Both local and remote videos

The possible causes are as follows:

- The local network is in poor condition or interrupted.
- If the network of one user is faulty, the video of this user cannot be displayed.
- The user disables the video.

If the problem persists, **submit a service ticket** to contact Huawei Cloud technical support.

6.5 Why Can't I Use the Camera?

Perform the following operations to locate the fault:

- Check whether the permission for using cameras is enabled. Both Android and iOS support permission management. Check the permissions in system settings. Permission management is also available for some Android security software.
- Check whether other apps are using the camera. If yes, disable the camera, restart your mobile phone, and then enable the camera again.
- Check whether the camera hardware malfunctions. Open the pre-installed video shooting app and check whether video can be recorded.

If the problem persists, you are advised to **submit a service ticket** for Huawei Cloud technical support.

6.6 Why Can't I Join Another Room?

The possible causes are as follows:

- There is more than one **Joiner** in different rooms at the same time.
- You try to join more than four rooms at a time, or the room IDs are not unique.
- The remote audio mode (**setRemoteAudioMode**) is set to **HWRtcRemoteAudioTopOfAll**.

6.7 What Do I Do If No Recording File Is Generated After SparkRTC Enables Cloud Recording?

Recording files may fail to be stored in OBS buckets due to the cloud resource storage authorization, freezing, or encryption of OBS. Perform the following operations to locate the fault:

- Check whether SparkRTC is authorized to store recording files in OBS buckets.
- Check whether the OBS bucket is in arrears.
- Check whether the OBS bucket is encrypted.

6.8 Why Can't I Join a Room?

Check whether access authentication has been performed. SparkRTC requires that all applications be authenticated before accessing the system. For details about the authentication method, see **Access Authentication**.

