SparkRTC

Service Overview

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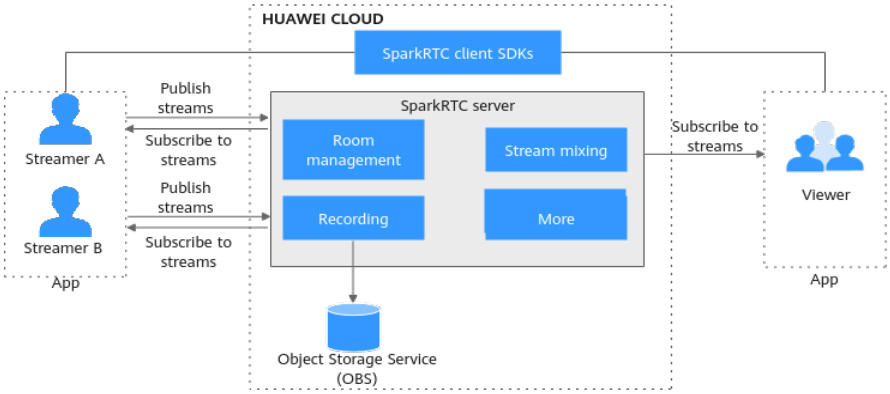
1 What Is SparkRTC?

SparkRTC, built on Huawei's long-established video technologies and expertise, delivers a high-concurrency and high-definition (HD) 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.



- SparkRTC client SDKs for web are provided for you to quickly integrate and develop functions such as audio and video calls and interactive livestreaming.
- SparkRTC server receives co-hosting requests, instructs hosts to join the room with real-time audio and video streams, and combines and records the streams published by streamers.

Accessing SparkRTC

You can access SparkRTC from the web-based management console and by calling application programming interfaces (APIs) based on HTTPS.

- Using the management console

  Log in to the management console to access SparkRTC.

  - If you have registered an account, log in to the management console and choose **Media Services** > **SparkRTC** from the service list.
  - If you have not registered an account, **register** one.
- Calling APIs

Call APIs to integrate SparkRTC into a third-party system for secondary development. For details, see the **Server API Reference**.

2 Product Advantages

Global Network

SparkRTC has a high-quality and large-scale real-time communication network across the globe. The in-house scheduling algorithm allows SparkRTC to efficiently schedule abundant nodes on the network, ensuring that the average E2E latency is less than 200 ms.

Fewer Freezes

With Huawei's 30 years of experience, SparkRTC is built on the latest audio and video codec technologies for optimum network quality as well as smooth audio and video experience, even when packet loss rates of audio and video reach 80% and 50%, respectively.

High-Quality Audio

Robust 3A algorithm, intelligent noise reduction, echo cancellation, intelligent feedback suppression, and 48 kHz sampling provide crystal clear audio quality.

High-Quality Video

1080p video, H.265, and perceptual encoding technologies reduce the bitrate by 30% to 40%. SCC screen sharing encoding retains high-fidelity colors for image and text.

Service Reliability

Annual service availability is at 99.99%. Full-link protection and E2E encryption secure your personal data.

3 Scenarios

Online Education

SparkRTC is applicable to online learning and interaction between teachers and students. It provides SDKs with HD, low-latency, and high-concurrency livestreaming, which can be quickly deployed on online education video platforms.

Entertainment and Social Interactions

SparkRTC is an ideal option in livestreaming scenarios of Internet celebrities, enterprises, entertainment, and gaming. It provides powerful real-time media processing capabilities and builds an end-to-end audio and video interaction solution for customers and partners.

Livestreaming E-commerce

SparkRTC is useful for improving the interaction experience and e-commerce customer conversion rate.

Video Conferencing

SparkRTC is perfect for low-latency video conferencing. It provides stable, smooth, and economical HD video conferencing, which features super packet loss concealment (PLC) and low latency.

4 Functions

HUAWEI CLOUD SparkRTC provides functions such as video calls and interactive co-hosting, as shown in **Table 4-1**.

Table 4-1 Functions

Function	Description	Scenario
Video call	One-to-one or multi-party audio call with 720p or higher HD quality. Up to 2000 users can join a room and up to 500 users can interact with each other at the same time.	One-to-one video call, video conferencing, online diagnosis, multi-participant video chat, video customer service, video and audio recording, and online claim settlement
Audio call	One-to-one or multi-participant audio call. Up to 2000 users can join a room and up to 500 users can speak at the same time.	One-to-one audio call, multi-participant audio call, audio chat, audio conferencing, and voice customer service
Interactive video livestreaming	Co-hosting and cross-room streamer competition are supported.	Ultra-low latency livestreaming, super large class, streamer competition, remote training, and large-scale conferencing
Interactive audio livestreaming	Co-hosting and cross-room streamer competition are supported.	Low latency livestreaming, co-hosting, karaoke, and FM broadcasting
High-quality audio	48 kHz sampling rate	Audio call, video call, interactive livestreaming, high-quality FM, music teaching, karaoke room, and online class

Function	Description	Scenario
High-quality image	720p and 1080p HD video	Video call, interactive livestreaming, and online class
Role switch	Users can switch their roles in a room quickly and smoothly.	Interactive livestreaming and online class
Joining multiple rooms	A streamer can join multiple rooms for competition.	Live show, co-hosting competition, and super large class
Screen sharing	Desktop, window, and rectangular area sharing	Interactive class, video conferencing, and remote assistance
3A processing	The industry-leading 3A algorithms, acoustic echo cancellation (AEC), automatic noise suppression (ANS), and automatic gain control (AGC), provide crystal clear audio quality.	All audio scenarios
Volume callback	SparkRTC provides volume values, which can be displayed as an animated sound wave or prompt.	Audio call, video call, voice chat room, FM radio, karaoke room, and voice detection
Cross-room co-hosting	Streamers can interact with each other across rooms. Viewers can smoothly switch their roles to hosts to speak.	Co-teaching and streamer competition
Interactive co-hosting	Viewers can smoothly switch their roles to hosts to speak.	Small class and interactive livestreaming
Cloud recording	Single and mixed streams are recorded and recordings are stored in OBS or hosted in VOD.	Storage, recording review, and video and audio recording
Audio mixing	Local or online audio files can be mixed with a user's voice in a room. The mixed audio can be sent to other users in the room.	Online education, chat room, and online chorus
Livestreaming via CDN	Audio and video streams in a room are mixed and transcoded to RTMP video streams, which are pushed to Live via Content Delivery Network (CDN) for livestreaming.	Interactive livestreaming and large-scale conferencing

Function	Description	Scenario
Network detection	The network detection API is used to determine or predict whether the network condition of a user is good.	-
Custom video data	Custom video sources and renderers are supported. Non-camera video sources, such as video files, external devices, and third-party custom data sources, can be used.	Beauty customization, data source customization, multi-device management, video recognition, and image processing
Custom audio data	Developers can collect audio callback data, process raw data, and perform custom operations, such as connecting to non-standard devices and audio files.	Non-standard device access, custom audio effect, voice processing, and voice recognition
Platform compatibility	The web platform is supported. For details about the system requirements, see **Constraints**.	-

5 Constraints

Before using SparkRTC, understand the following constraints.

Table 5-1 Constraints

Constraint	Description
App creation	If this is the first time you use Huawei Cloud SparkRTC, **submit a service ticket** to contact Huawei Cloud technical service for more information.
Maximum number of users in a room	500
Web SDK	• Mainstream browsers such as Chrome, Firefox, Safari, and Opera are supported. For details, see . • Currently, it cannot be used for applet integration.

6 Concepts

App ID

The app ID is a unique identifier used by SparkRTC to identify apps. It is automatically generated when an app is created on the SparkRTC console.

Room

A room is an audio and video space where users can receive real-time audio and video from each other.

- SparkRTC uses virtual rooms to isolate users.
- Only users in the same room can receive audio and video from each other.
- The naming rule of a room ID can be customized. A room ID contains 64 characters, including letters, digits, underscores (_), and hyphens (-).

User ID

Unique identifier used by SparkRTC to identify users in an app.

- A user ID is a representation of a user in SparkRTC and can be customized.
- A user ID contains 64 characters, including letters, digits, underscores (_), and hyphens (-).

Role

Role of a user in a room. Various roles have 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 users of the joiner and player role.

Video Stream

Stream collected by cameras. SparkRTC can encode, receive, and send video streams of up to four resolutions (720p, 360p, 180p, and 90p) from the same source at the same time.

Presentation Stream

Sharing stream, which is the stream of the shared screen or program. The default resolution is 1080p. If the fluency is preferred, the presentation resolution will be changed to 720p.

Stream Mixing

Multiple pushed audio and video streams are combined into a single stream.

Dual Stream Mode

By default, only high-quality streams (720p) are encoded, received, and sent. In dual stream mode, high-quality streams (720p) and low-quality streams (360p) are both encoded, received, and sent.

Subscription

Operation that a user in a room receives audio and video streams from a remote user in the room.

Custom Collection and Rendering

By default, SparkRTC uses system audio and video devices to collect and render data. It also supports custom collection and rendering for screen recording playback, third-party beautification and special effects, and cloud gaming.

Packet Loss

Packet loss occurs when packets of real-time audio and video data traveling across a network fail to reach their destination.

Jitter

Variation in time delay of data packets transmitted continuously over a network connection

Freeze

Intermittent, unsmooth, or even frozen audio or video playback caused by poor network conditions or limited device performance during real-time audio and video transmission

Single Stream Recording

Audio and video streams of each user in a room are recorded as one file.

Mixed Stream Recording

Audio and video of multiple users in a room are mixed and recorded as one file.