

Tencent Real-Time Communication

Product Introduction

Product Documentation



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Tencent RTC Quickplay: Experience Ultimate Real-Time Audio and Video Interaction!

Product Introduction

Overview

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Leveraging Tencent's 21 years of experience in network and audio/video technologies, Tencent Real-Time Communication (TRTC) offers solutions for group audio/video calls and low-latency interactive live streaming. With TRTC, you can quickly develop cost-effective, low-latency, and high-quality interactive audio/video services.

Group audio/video call

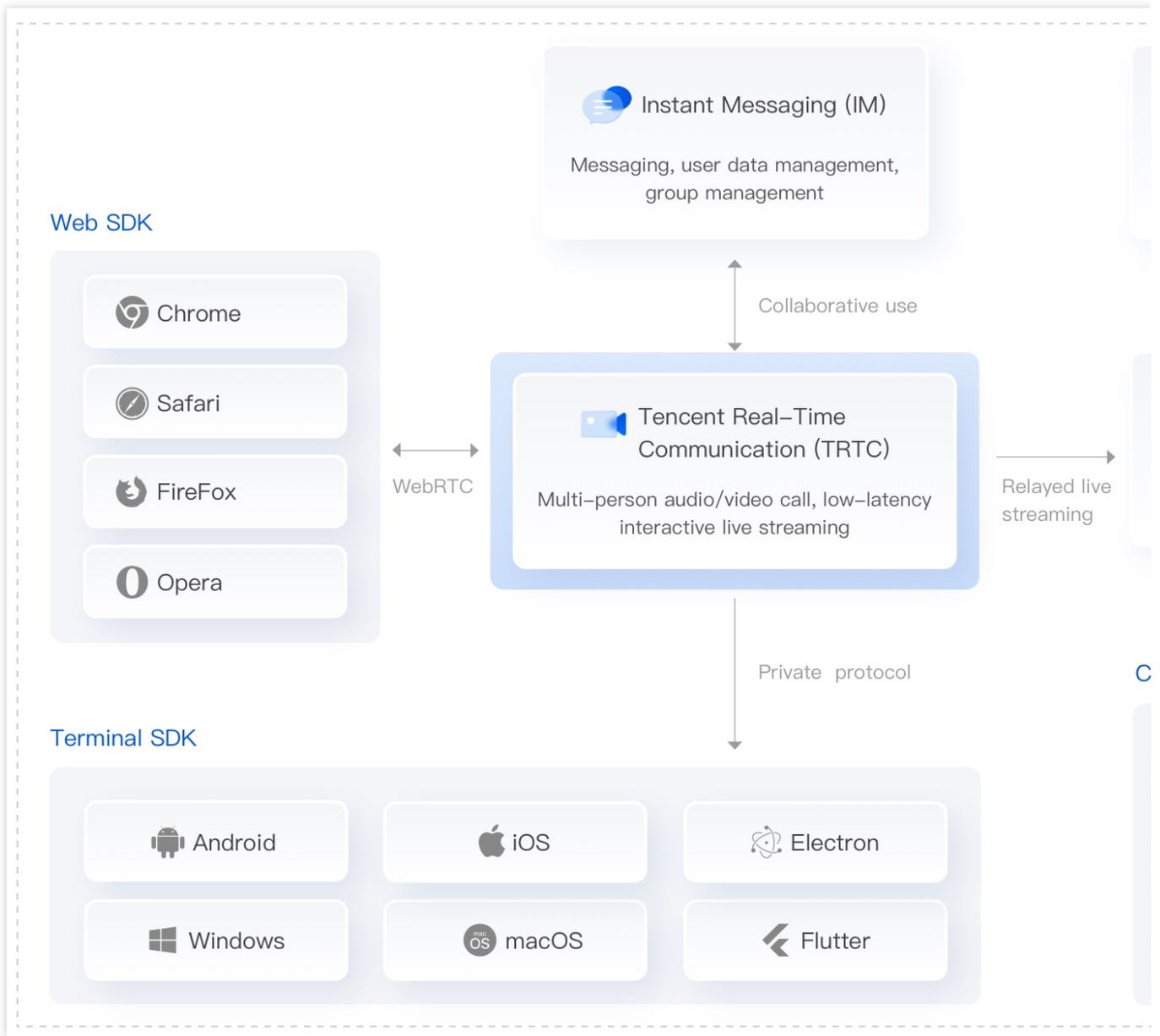
Built on Tencent Cloud's Direct Connect network, this solution allows global connection and offers client SDKs and cloud-based APIs for both mobile and desktop platforms. Users can also easily access TRTC services on webpages.

Low-latency interactive live streaming

This solution leverages Tencent Cloud's industry-leading network and audio/video technologies as well as high-quality node resources to help you build interactive live streaming services with minimal lag and latency below 1 second, taking live streaming into the era of CDN 2.0.

Product Architecture

TRTC offers cross-platform solutions for audio/video calls and low-latency interactive live streaming. With SDKs for mini programs, web, Android, iOS, Electron, Windows, macOS, and other platforms, you can quickly integrate TRTC services into your projects and connect to the TRTC backend. You can also combine TRTC with other Tencent Cloud products such as Instant Messaging (IM), Cloud Streaming Services (CSS), and Video on Demand (VOD) to explore more use cases. The figure below shows how TRTC works together with other products.



Supported Platforms

TRTC **supports a wide range of platforms**. Below is a list of the supported platforms and the development environments required.

| Platform | Environment Requirements |
|----------|---|
| iOS | iPhone and iPad on iOS 9.0 or above. Xcode 9.0+ A valid developer signature for your project. |
| Android | Android Studio 3.5+ |

| | |
|----------|---|
| | Android 4.1 (SDK API Level 16) or above. |
| Windows | Windows 7 or above. Visual Studio 2010 or above (2015 is recommended). .Net Framework 4.0 or above. |
| macOS | Xcode 9.0+ Mac on macOS X10.10+. A valid developer signature for your project. |
| Web | Desktop Chrome 56+, Edge 80+, Firefox 56+, Safari 11+. Mobile Android Chrome 56+, iOS Safari 11+. Fore more, please refer to Supported Platforms . |
| Electron | Windows 7 or above; macOS X10.10 or above. Electron 4.0.0 or above (the latest Electron SDK is recommended). |
| Flutter | iOS: iPhone and iPad on iOS 9.0 or above. Xcode 9.0+ A valid developer signature for your project. Android: Android Studio 3.5+. Android 4.1 (SDK API Level 16) or above. |

Concepts

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This document explains some of the basic concepts you may encounter while using TRTC services.

Application

In TRTC, business and projects are managed with **applications**. You can create different applications for your projects in the [TRTC console](#) to separate their data. Each Tencent Cloud account can create up to 100 TRTC applications.

SDKAppID

Tencent Cloud uses `SDKAppID` (application ID) to uniquely identify TRTC applications. It is generated automatically when you create an application in the [TRTC console](#). Applications that do not have the same `SDKAppID` cannot communicate data with each other.

UserID

UserID (user ID) uniquely identifies a user in a TRTC application.

`UserID` is a mapping of the user accounts of your project in Tencent Cloud. Normally, you can use a username as `UserID`.

`UserID` should preferably be 32 bytes or shorter. It can contain digits, underscores, and letters (case sensitive).

Room

A room is a space where users can receive each other's audio and video data in real time.

Rooms are virtual spaces TRTC uses to separate one user group from another.

Only users in the same room can receive each other's audio and video.

A user can be in only one room at a time. A user who is already in a room must exit the room before they can enter another.

notice

The first user who enters a room is the owner of the room. Room owners cannot close rooms manually.

In call modes, TRTC closes a room when all users leave the room.

In live streaming modes, if the last user who exits a room is an anchor, TRTC will close the room immediately; if the user is an audience member, TRTC will close the room after 10 minutes.

A user is removed from a room 90 seconds after an unexpected disconnection occurs. If all users are disconnected, the room is closed after 90 seconds. **The waiting time after a disconnection occurs is also billed.**

If a user attempts to enter a room that does not exist, TRTC creates a room automatically.

Room ID

`RoomID` (room number/ID) uniquely identifies a room in a TRTC application. It is a UInt32 number that you assign to a room and maintain. Its value range is 1-4294967295.

UserSig

`UserSig` (user signature) is a security signature designed by Tencent Cloud to authenticate user logins and verify that a user is real. It helps prevent attackers from accessing your Tencent Cloud account. For more information, see [FAQs > UserSig](#).

Push

Push is the operation in which a user uploads local audio/video data to the TRTC server.

Subscription

Subscription is the operation in which a user sends a request to the TRTC server to pull the audio/video data of a specified user.

Role

In TRTC, users can have either of two roles: **anchor** (`TRTCRoleAnchor`) and **audience** (`TRTCRoleAudience`).

An anchor can push local audio/video data to the server. They can also subscribe to and play back the audio/video data of other anchors.

Audience members **can only** subscribe to and play back the audio/video data of anchors.

In call modes, all users in a room have the anchor role. In live streaming modes, users in a room can be either anchors or audience members. A user can switch roles as needed.

CDN Live Watching

CDN live watching is also known as CDN relayed live streaming. TRTC uses relayed transcoding clusters to convert its UDP audio/video streams into RTMP streams in the cloud. The streams are then pushed to a standard live streaming system and distributed through CDNs to the audience. For details, see [Relay to CDN](#).

On-cloud recording

TRTC leverages the capabilities of [CSS](#) to record entire audio/video calls in the cloud. Recording files are securely saved in real time to [VOD](#). For details, see [On-Cloud Recording](#).

On-Cloud MixTranscoding

In scenarios such as **CDN live watching** and **on-cloud recording**, you may need to mix multiple audio/video streams in a TRTC room into one stream. This can be achieved using TRTC's stream mixing and transcoding MCU cluster. The MCU cluster can mix multiple audio/video streams as needed and distribute the mixed stream to live streaming CDNs and the on-cloud recording system.

Dumb terminal

A user entering a room on a dumb terminal will not be detected by the SDK, and remote users will not receive notifications when the user enters or exits the room.

Features

Last updated : 2024-07-05 20:13:14

Basic Features

| Feature | Description | Use Cases | Billing |
|-----------------------------|--|---|---|
| Video call | <p>720p/1080p one-to-one or group video calls</p> <p>Each room allows up to 300 concurrent users, and up to 50 of them can turn their cameras on at the same time.</p> | <p>One-to-one video calls, video conferences with up to 300 participants, online medical consultation, video chat, video customer service, video interviews, dual-channel recording, online insurance claim settlement, and video-based party games</p> | <p>Billing of TRTC Basic Services</p> |
| Audio call | <p>One-to-one or group audio calls, which support 48 kHz sampling and dual channels</p> <p>Each room allows up to 300 concurrent users, and up to 50 of them can keep their mics on at the same time.</p> | <p>One-to-one or group audio calls, voice chat, audio conferences, audio customer service, and audio-based party games</p> | |
| Interactive video streaming | <p>Video communication between anchors and audience members</p> <p>Cross-room anchor communication</p> <p>Smooth mic on/off without waiting; anchor latency less than 300 ms</p> <p>No upper limit on the cumulative number of co-anchoring users in a</p> | <p>Low-latency live video streaming, interactive classrooms with up to 100,000 participants, live video</p> | |

| | | | |
|-----------------------------|--|--|---|
| | <p>room; up to 50 users can keep their mics on at the same time.</p> <p>Streaming to up to 100,000 concurrent users and a playback latency as low as 1,000 ms in the low-latency live streaming mode</p> <p>No upper limit on audience size in the CDN live streaming mode</p> | <p>competitions, video dating, remote training, large-scale conferences</p> | |
| Interactive audio streaming | <p>Audio communication between anchors and audience members</p> <p>Cross-room anchor communication</p> <p>Smooth mic on/off without waiting; anchor latency less than 300 ms</p> <p>No upper limit on the cumulative number of co-anchoring users in a room; up to 50 users can keep their mics on at the same time.</p> <p>Streaming to up to 100,000 concurrent users and a playback latency as low as 1,000 ms in the low-latency live streaming mode</p> <p>No upper limit on audience size in the CDN live streaming mode</p> | <p>Low-latency live audio streaming, live audio co-anchoring, live audio competitions, audio chat rooms, audio dating, karaoke rooms, FM radio</p> | |
| On-Cloud Recording | <p>Uses a recording MCU cluster to record the upstream audio and video streams in a room either separately or after mixing. Recording files are saved securely and in real time to VOD.</p> | <p>Dual-channel recording, archiving, compliance</p> | <p>On-cloud recording is a value-added service and incurs additional fees.</p> |
| On-Cloud MixTranscoding | <p>Uses an MCU cluster to mix and transcode the audio and video streams in a room and publishes the mixed stream to CSS for on-cloud recording or CDN playback.</p> | <p>Stream mixing, recording format conversion</p> | <p>On-Cloud MixTranscoding is a value-added service and incurs additional fees.</p> |
| Publishing to CSS CDN | <p>Uses a relay and transcoding cluster to convert UDP streams into RTMP streams in the cloud, pushes the streams to CSS, and distributes them via CDNs.</p> | <p>Interactive live streaming, live sharing, large-scale conferencing, live stream viewing by remote users</p> | <p>Relay to CDN is a value-added service and is charged by CSS. For details, see Relay to CDN > Billing.</p> |

Advanced Features (Paid)

| Feature | Description | Use Cases | Billing |
|--|---|---|--|
| AI noise suppression | Removes intermittent noises that traditional noise suppression technologies cannot handle, such as cough, sneeze, and car horn. If Chat features are used, noise suppression can work for both TRTC calls and local recordings. | Audio call, video call, interactive live streaming, audio chat room, online class | Buy a TRTC monthly package to unlock these features. |
| Less stutter under poor network conditions | Reduces stutter rate and loading time under poor network conditions. | Audio call, video call, interactive live streaming, audio chat room, online class | |
| RTMP Streaming with TRTC | Publishes and plays streams using RTMP. | Online class, sports event streaming | |
| 3D spatial audio | Delivers different audio effects according to the face orientation of a virtual anchor and the orientation and location of audio sources to enable a spatial and life-like sound experience. | Audio chat room, karaoke room, FM radio, virtual concert, online gaming | |
| Scaled video coding | Leverages the O264RT coding technology of Tencent Media Lab to reduce loading time, notably cut bandwidth consumption, and improve compatibility with devices. | Video call, online class, interactive live streaming | |
| Region of interest coding | Reduces CPU usage and delivers a video experience with the lowest possible latency and highest possible quality in most scenarios. | Video call, online class, interactive live streaming | |
| High video resolution | The anchor can publish videos and share the screen at a resolution of 2K/4K (only 2K+ is supported on PC and web currently). | Video call, online class, interactive live streaming | |
| Voice changing effects | Uses acoustic algorithms to change the sound of a speaker's voice. | Audio chat room, karaoke room, FM | |

| | | | |
|--------------------------|---|--|--|
| | | radio, virtual concert, online games, etc. | |
| Virtual Background (Web) | Based on portrait contour recognition technology, it achieves precise portrait segmentation, supporting background blurring and background replacement. Currently available only on the Web , more features and platform support are coming soon. | Video calls, interactive live streaming, online meetings, online classes, etc. | |

Other Advanced Features

| Feature | Description | Use Cases | Billing |
|-------------------------------|---|--|--|
| Anchor-audience communication | Audience members can turn their mics to communicate with the anchor. The mic-on/off process is smooth and requires no waiting. | Interactive live streaming, online class, chat room | The feature itself is free, but audio/video call or live streaming durations will be generated when you use this feature, so TRTC basic service fees will be incurred. |
| Cross-room communication | Anchors from different rooms can communicate with each other while audience members watch. | Showroom streaming, cross-room match, cross-room class | The feature itself is free, but audio/video call or live streaming durations will be generated when you use this feature, so TRTC basic service fees will be incurred. |
| Screen sharing | The anchor can share the desktop, a window (for example, a Microsoft PowerPoint window), or a portion of the desktop to others. | Online class, slide sharing, remote assistance | The feature itself is free, but audio/video call or live streaming durations will be generated when you use this feature, so |

| | | | |
|------------------------|---|--|--|
| | | | TRTC basic service fees will be incurred. |
| Local server recording | Server-side recording relies on the Linux SDK, which is not commercially available yet. If you have questions about the SDK or want to use it, please contact us at colleenyu@tencent.com. | Dual-channel recording, archiving, compliance | The feature itself is free, but audio/video call or live streaming durations will be generated when you use this feature, so TRTC basic service fees will be incurred. |
| High sound quality | 48 kHz sample rate, end-to-end 192 Kbps bitrate, and dual channels for a clear and immersive audio interaction experience | Audio call, video call, interactive live streaming, audio chat room, high-audio-quality FM radio, music class, karaoke, online class | Free |
| High video quality | 720/1080p video quality | Video call, interactive live streaming, online class | Free |
| 3A processing | Leverages the industry-leading 3A (acoustic echo cancellation, active noise suppression, automatic gain control) technologies of Tencent Ethereal Audio Lab to deliver high audio quality even when multiple people speak at the same time or in the presence of background noises. | All audio scenarios | Free |
| Basic beauty filters | Basic beautification effects, including skin brightening, skin smoothing, rosy skin, and basic filters | Video call, interactive live streaming, online class | Free |
| BGM | The anchor can play an audio file in MP3, AAC, or WAV format as background music during a call or live stream. | Audio call, video call, interactive live streaming, online class, audio chat room, karaoke room, FM radio | Free |
| Audio effects | Audio effects such as applauding, cheering, whistling, and booing | Audio call, video call, interactive live | Free |

| | | | |
|---------------------------|---|--|------|
| | | streaming, audio chat room, karaoke, FM radio | |
| Publishing computer audio | The anchor can publish the audio they play locally, for example, the music played by QQ Music on their computer, to remote users. | Interactive live streaming, online class, audio chat room, FM radio | Free |
| Voice changing | Voice changing effects such as little girl, middle-aged man, and metal | Audio call, video call, interactive live streaming, audio chat room, karaoke, FM radio | Free |
| Reverb | Reverb effects such as karaoke, small room, hall, and shower room | Audio call, video call, interactive live streaming, audio chat room, karaoke, FM radio | Free |
| Audio volume callback | Callback of audio volumes, which you can use to display audio waveforms or show volume reminders | Audio call, video call, audio chat room, FM radio, karaoke, and speech detection | Free |
| In-ear monitoring | Records local audio and plays it back in the local user's earphones, usually for detection of speech errors or pitch control during singing | Interactive live streaming, showroom streaming, karaoke | Free |
| Custom audio data | Callback of raw audio captured from a non-standard external device or a local audio file for custom processing. | Non-standard device connection, custom audio effect, speech processing, speech recognition | Free |
| Custom video data | Custom video sources (such as video files, external devices, and third-party data sources) and renderers | Custom beauty filters, custom data sources, multi-device management, video recognition, image processing | Free |
| SEI message | Embeds custom information such as lyrics and questions as SEI frames into published video streams. | Karaoke, live quiz, interactive live streaming | Free |

Extended Features

Note
Extended features are provided by other Tencent Cloud products and are charged according to the billing standards of the corresponding products.

| Feature | Description | Use Cases | Billing |
|-------------------|--|---|--|
| Chat | You can use the capabilities of Tencent Cloud Chat, including one-to-one chat, group chat, and chat rooms with no upper limit on the number of users, to implement features such as chat, comments, on-screen comments, gift sending, and liking. You can also use Chat to implement signaling-based interaction, call invitations, and user counting. | Online customer service, interactive live streaming, interactive classroom, remote training | These features incur Chat fees. For details, see Pricing . |
| AI beauty filters | Diverse effects enabled by face recognition technologies, such as AI-based beautification, makeup effects, facial feature adjustment, and green screen keying | Video call, interactive live streaming, showroom streaming | This feature incurs Tencent Effect SDK fees. |

Strengths

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Cross-platform global connection

TRTC is a cross-platform solution compatible with more than 30,000 device models. It provides client SDKs and Tencent Cloud APIs for both mobile and desktop platforms including iOS, Android, Windows, macOS, and web.

Easy and quick integration

You can run a TRTC demo and integrate basic TRTC features into your project with a few lines of simple code. In as little as one minute, you can build a high quality real-time audio/video communication product from scratch featuring low latency and low stutter rate. For detailed directions, see [API Examples](#) and [SDK Quick Integration](#).

Scenario-specific components

TRTC provides a rich set of components to help you quickly implement features such as audio chat, conferencing, interactive live streaming, and interactive teaching. For detailed directions, see [Building a demo](#).

Low latency

TRTC offers reliable and secure network connection across the globe. It uses Tencent Cloud's self-developed multi-level addressing algorithm and can connect to nodes across the entire network. Abundant high-bandwidth resources and globally-distributed edge servers allow it to maintain **an average end-to-end latency of below 300 ms** across different countries and regions.

Low stutter rate

TRTC reduces stutter through intelligent QoS control and optimized encoding. It guarantees high-quality, smooth, and stable audio/video communication even under poor network conditions (**packet loss over 80%** and **network jitter over 1,000 ms**).

High video/audio quality

TRTC supports **720p and 1080p** video and allows video calls even under a packet loss rate of 70%. It supports **48 kHz** audio and uses the industry-leading 3A technologies of Tencent Ethereal Audio Lab to remove echo and howling. End-to-end 128 Kbps bitrate and dual channels ensure a clear and immersive audio interaction experience.

Use Cases

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Tencent Real-Time Communication (TRTC) offers two main solutions: low-latency interactive live streaming and audio/video call. Its capabilities include low-latency live streaming, real-time recording, screen sharing, beauty filters, stereo sound, and relay to CDN. It is ideal for co-anchoring, cross-room communication, radio, karaoke, small/big online classes, audio chat, video chat, online conferencing, and other applications. This document describes the main use cases of TRTC's interactive live streaming and audio/video call solutions.

Interactive Audio Streaming

Audio chat room

Up to 50 speakers at a time; smooth mic on/off with a latency below 300 ms; diverse voice changing effects, ambient sound effects, and reverb effects for a rich chat experience; instant messaging features such as public chat, private chat, group chat, liking, and gifting provided by Tencent Cloud Chat. You can use our TUIVoiceRoom component to build your audio chat room application with ease and at minimal development costs. For more information on how to use it, see [Interactive Audio Streaming](#).

Radio

Stereo audio at a sample rate of 48 kHz and a bitrate of 192 Kbps; supports use of local files in MP3, AAC, WAV, and other formats as the background music; diverse voice changing effects (middle-aged man, little girl, and more) to make your radio more entertaining. You can use our TUIVoiceRoom component to build your radio application with ease and at minimal development costs. For more information on how to use it, see [Interactive Audio Streaming](#).

Online karaoke

Stereo audio at a sample rate of 48 kHz and a bitrate of 128 Kbps for a studio-like karaoke experience; duet latency below 300 ms; multiple synchronization mechanisms such as message pass-through and timestamp to ensure excellent syncing of the instrumental track, vocal, and lyrics; in-ear monitoring for pitch control.

Interactive Video Streaming

Showroom streaming

Cross-room and same-room communication with latency below 300 ms; smooth mic on/off; smart beauty filters. You can use our TUILiveRoom component to build your showroom streaming application with ease and at minimal development costs. For more information on how to use it, see [Interactive Video Streaming](#).

Interactive big class

Streaming to 100,000 students at the same time with a latency below 300 ms; teacher-student co-anchoring; smooth mic-on/off; screen sharing, interactive whiteboard, recording and playback, and other features for a richer and more interactive online learning/teaching experience. You can use our component to build your interactive class application with ease and at minimal development costs. For more information on how to use the component, see [Real-Time Interactive Teaching](#).

Interactive small class

Small classes of different sizes (**1-to-1**, **1-to-2**, **1-to-6**, **1-to-32**, and more), teacher-student communication with a latency below 300 ms; screen sharing, courseware sharing, interactive whiteboard, recording and playback, and other features for a richer and more interactive online learning/teaching experience.

Live quiz

Low-latency, high-concurrency live streaming; multiple synchronization mechanisms such as message pass-through, timestamp, and signaling channels to ensure excellent syncing of the audio, video, and quiz; high-concurrency instant messaging; quizzes on live streams, result statistics collection, and multi-person co-anchoring for a more engaging online quiz experience; real-time keyword and hint filtering to improve user experience and reduce the risks of non-compliance.

Audio Call

Group audio call

Each call allows at most 300 participants, and up to 50 can keep their mics on at the same time; 48 kHz sample rate, 128 Kbps bitrate, and powerful 3A processing technologies for a smooth and high-quality audio call experience. You can use our TUICalling component to build your group audio call application with ease and at minimal development costs. For more information on how to use it, see [Audio Call](#).

One-to-one audio call

A latency below 300 ms; smooth and stable audio call under a packet loss rate over 80% and network jitter over 1,000 ms; a rich set of signaling management APIs provided by Tencent Cloud Chat to meet the needs of different audio call scenarios. You can use our TUICalling component to build your audio call application with ease and at minimal development costs. For more information on how to use it, see [Audio Call](#).

Party games

A latency below 300 ms; smooth and stable communication under a packet loss rate over 80% and network jitter over 1,000 ms; real-time network monitoring; audio device testing to ensure that all players can be heard, improving gaming experience.

Audio conferencing

Cross-platform compatibility with devices including mobile phones, PCs, and tablets; powerful 3A processing technologies to remove echo and howling and ensure smooth and clear conferencing; interactive whiteboard and file sharing for a more efficient conferencing experience.

Video Call

Group video call

Each room allows at most 300 participants, and up to 50 can keep their cameras on at the same time; 720p/1080p video quality; instant messaging, video on demand, recording, and porn detection features to meet the needs of different video call scenarios. You can use our TUICalling component to build your video call application with ease and at minimal development costs. For more information on how to use it, see [Audio/Video Call](#).

One-to-one video call

High-quality (720p/1080p) one-to-one video call; instant messaging, screen sharing, recording, interactive whiteboard, and other features to meet the needs of different video call scenarios. You can use our TUICalling component to build your video call application with ease and at minimal development costs. For more information on how to use it, see [Audio/Video Call](#).

Online conferencing

Screen sharing, file sharing, and interactive whiteboard for a more efficient conferencing experience; text/image chat enabled by Tencent Cloud Chat. You can use our TUIRoom component to build your conferencing application with ease and at minimal development costs. For more information on how to use it, see [Audio/Video Conference](#).

Online healthcare

1080p video call for multiple physicians and patients at a time; flexible focus adjustment, medical record/medical image sharing, screen sharing, and instant messaging for an efficient online medical consultation experience comparable to traditional healthcare.

Video customer service

A latency below 300 ms; smooth and stable video call under a packet loss rate over 70% and network jitter over 1,000 ms; communication across mobile, desktop, and web; customer service accessible anytime and anywhere; recording and playback to improve service quality.

Financial transaction recording

On-cloud or server-side local recording in real time; file storage, playback, and download capabilities to help you ensure compliance; Tencent's 21-year experience in data security to provide robust protection for your data.

Performance Statistics

Last updated : 2024-05-21 15:05:29

This document analyzes TRTC's performance in terms of audio/video quality, latency, smoothness, stability, CPU usage, memory usage, battery consumption, heating, and other key indicators in tests under **normal** and **poor** network conditions and in **different application scenarios** (one-to-one, one-to-many, etc.).

Performance Under Normal and Poor Network Conditions

Scenario

Video call, interactive live streaming, and audio call

Parameter configuration

Video call:

| Parameter | Value |
|------------|-----------|
| Resolution | 368 x 640 |
| Bitrate | 400 Kbps |
| Frame rate | 15 |

Interactive live streaming:

| Parameter | Value |
|------------|------------|
| Resolution | 720 x 1280 |
| Bitrate | 1200 Kbps |
| Frame rate | 15 |

Poor network tolerance test

The TRTC SDK was tested for its tolerance to different bad network conditions.

| Poor Network Tolerance (iPhone XR to Xiaomi Mi 9) | | | |
|--|--|-------------------------|-------------------------|
| | Scenario | Live Streaming | Video call |
| Upstream | Maximum tolerable packet loss | 55% loss | 65% loss |
| | Maximum tolerable jitter | 1200ms | 1700ms |
| | Minimum bandwidth for smooth call | 500kbps | 250kbps |
| | Maximum tolerable packet loss + jitter | 20% loss + 300ms | 40% loss + 700ms |
| | Maximum tolerable packet loss + latency | 20% loss + 350ms | 40% loss + 700ms |
| Downstream | Maximum tolerable packet loss | 70% loss | 65% loss |
| | Maximum tolerable jitter | 1700ms | 1600ms |
| | Maximum tolerable packet loss + jitter | 40% loss + 800ms | 40% loss + 700ms |
| | Maximum tolerable packet loss + latency | 40% loss + 650ms | 40% loss + 600ms |

Note

For the metrics used to measure tolerance to poor network conditions, please see [Appendix 1: Network Metrics](#).

Audio MOS under poor network conditions

TRTC can deliver relatively high-quality audio and low latency under poor network conditions.

The table below lists TRTC's performance and mean opinion score (MOS) under different poor network conditions.

| Audio MOS Under Poor Network Conditions | | | | |
|--|-------------------------------|--|--------------|---------------|
| Use cases | | Android to Android (Xiaomi Mi 8 to Xiaomi Mi 9) | | (iPhor |
| | | score | ms | score |
| Upstream/ Downstream | Normal | 4.75 | 186 | 4.74 |
| Upstream | 75% loss | 3.82 | 570.7 | 3.82 |
| | 2000 jitter | 4.28 | 1362 | 4.32 |
| | 55% loss + 1200 jitter | 3.59 | 1570 | 3.57 |
| Downstream | 70% loss | 4.03 | 552 | 4.08 |
| | 2000 jitter | 3.53 | 1584 | 3.68 |
| | 50% loss + 900 jitter | 4.04 | 1392 | 3.93 |

Client SDK Performance

Tested devices

| Device | Processor | Memory |
|------------------|-----------------------------------|--------|
| Android device 1 | Qualcomm Snapdragon 835 - 8 cores | 6 GB |
| Android device 2 | Kirin 980 - 8 cores | 8 GB |
| iOS device 1 | Apple A8 - 2 cores | 1 GB |
| iOS device 2 | Apple A13 - 6 cores | 4 GB |

Parameter configuration

| Parameter | Value |
|------------|-----------|
| Resolution | 240 x 320 |
| Bitrate | 100 Kbps |

| | |
|------------|----|
| Frame rate | 15 |
|------------|----|

Test scheme

Scenario: one-to-one, one-to-two, one-to-four, one-to-eight

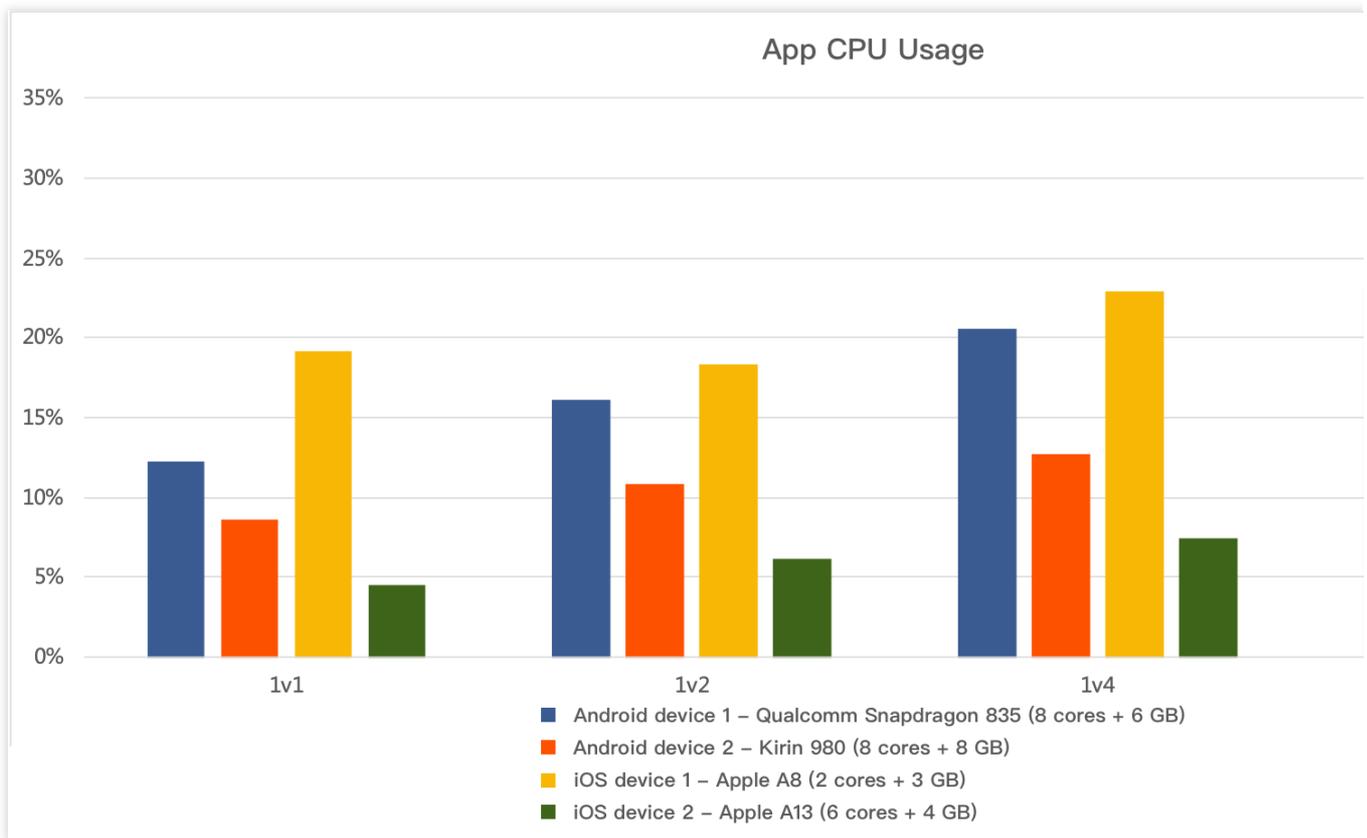
Duration: 30 min for each scenario

Method: a Linux robot is used to simulate streaming in one-to-many scenarios. Each device is tested independently.

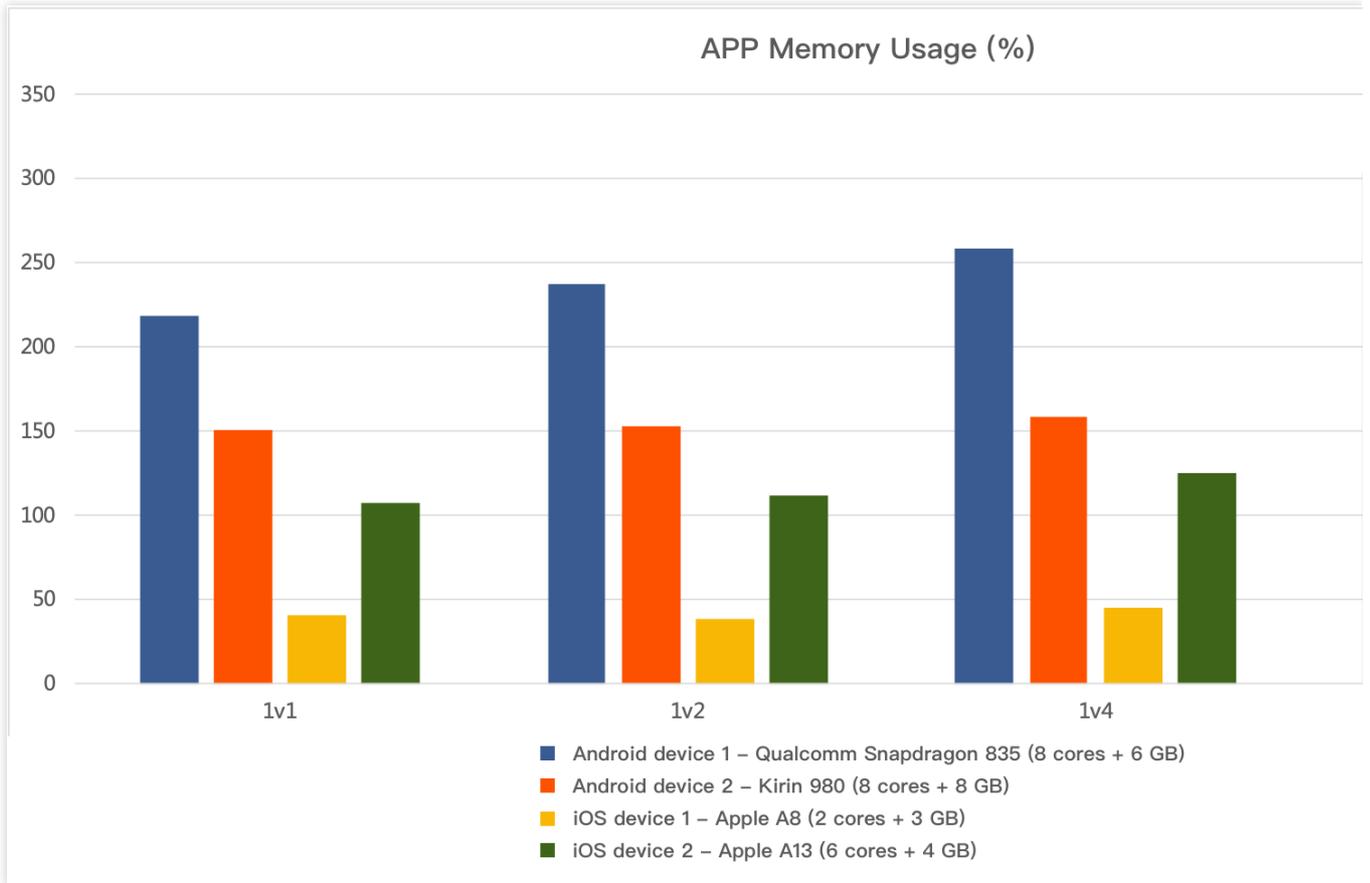
Test result

The TRTC SDK performs well in terms of CPU usage, memory usage, heating, and battery consumption. It uses a small amount of hardware resources but provides quality audio/video services.

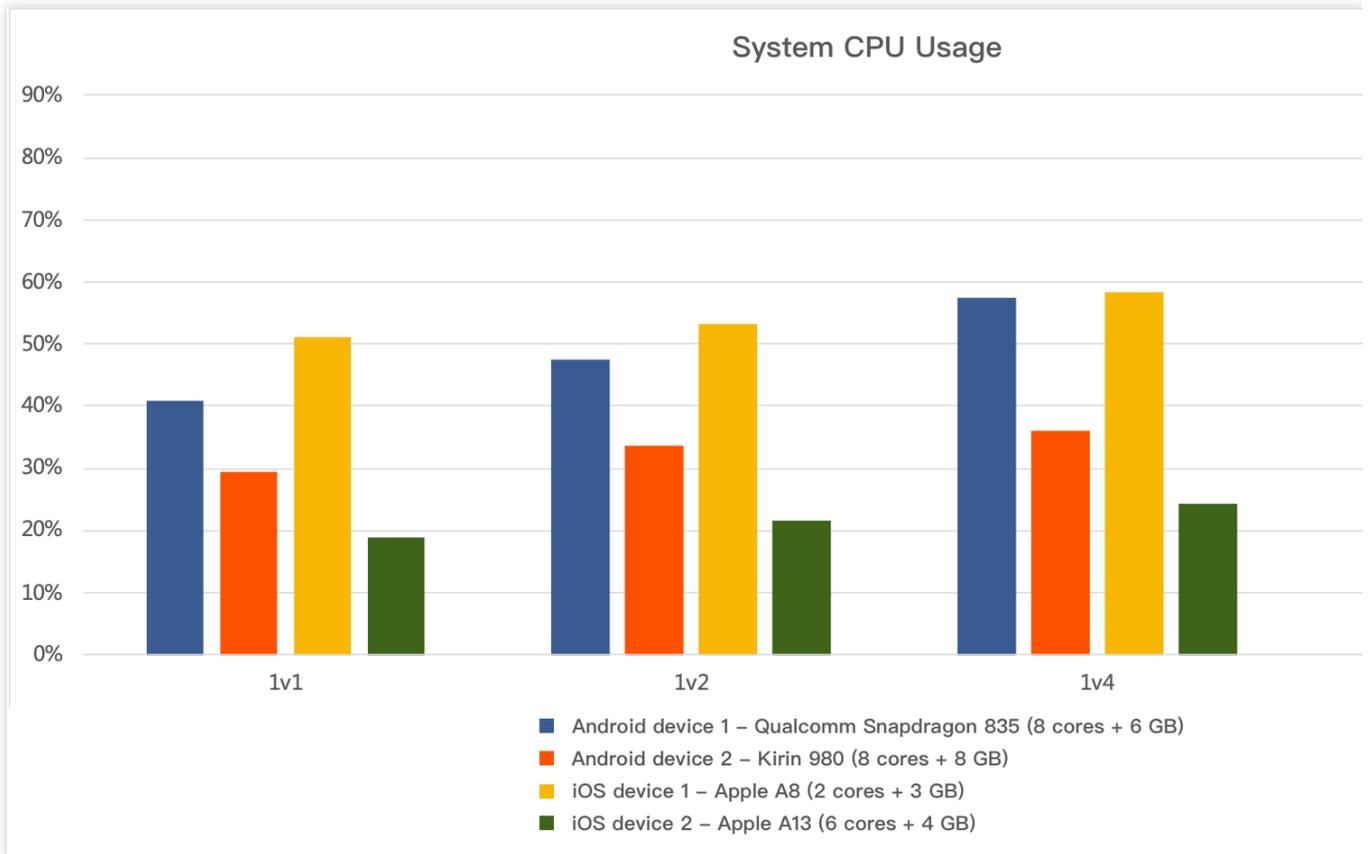
App CPU usage:



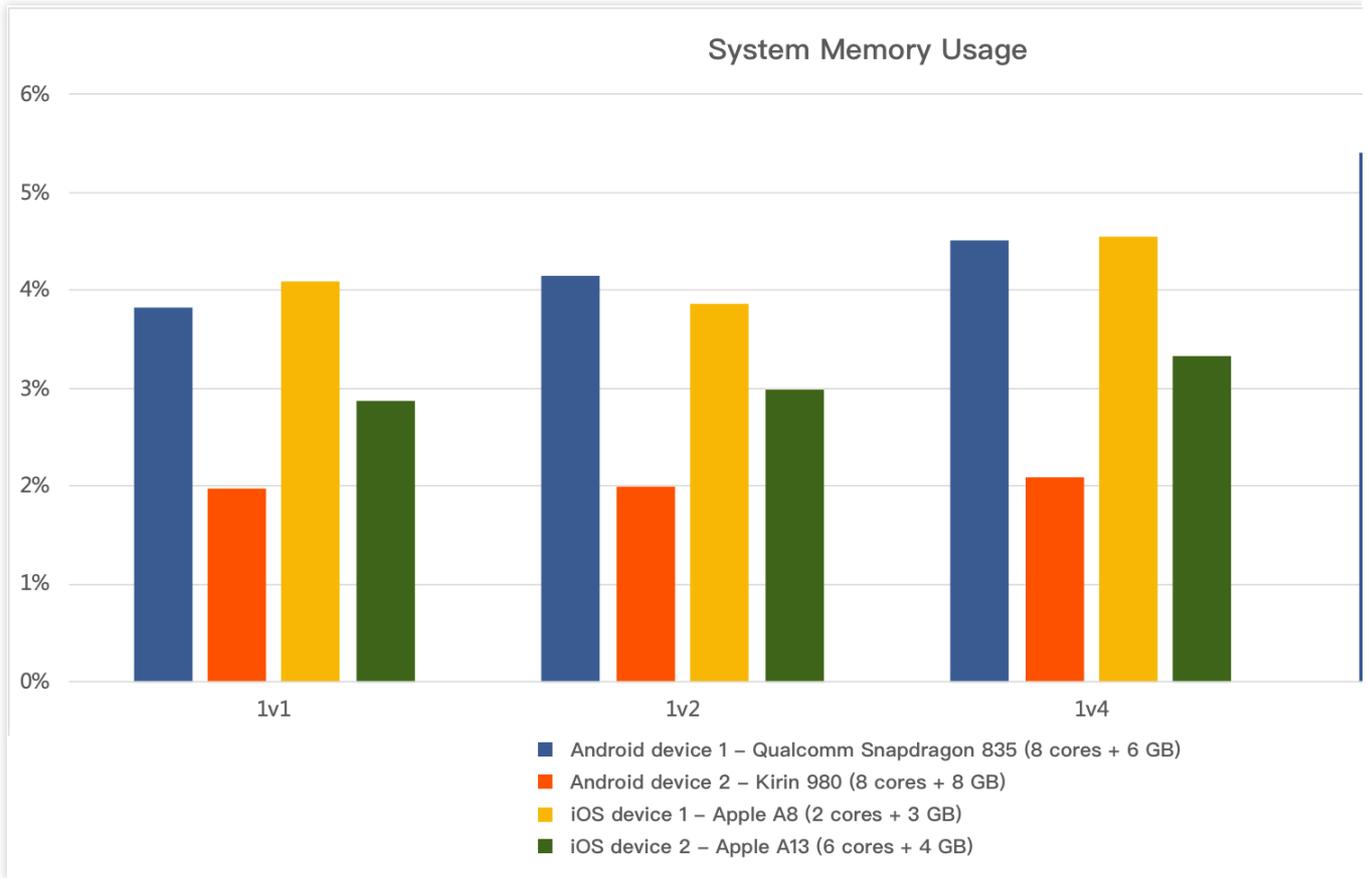
App memory usage:



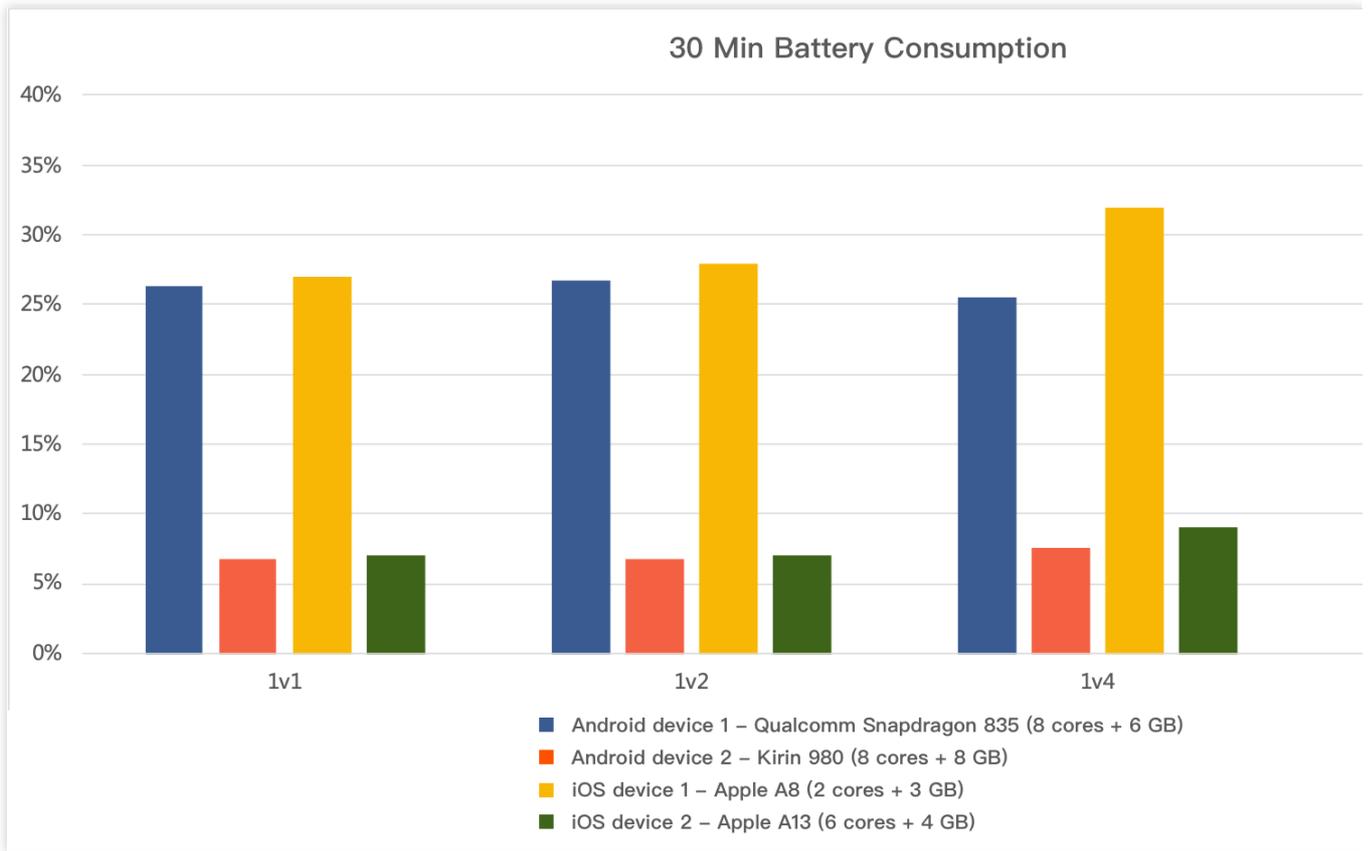
System CPU usage



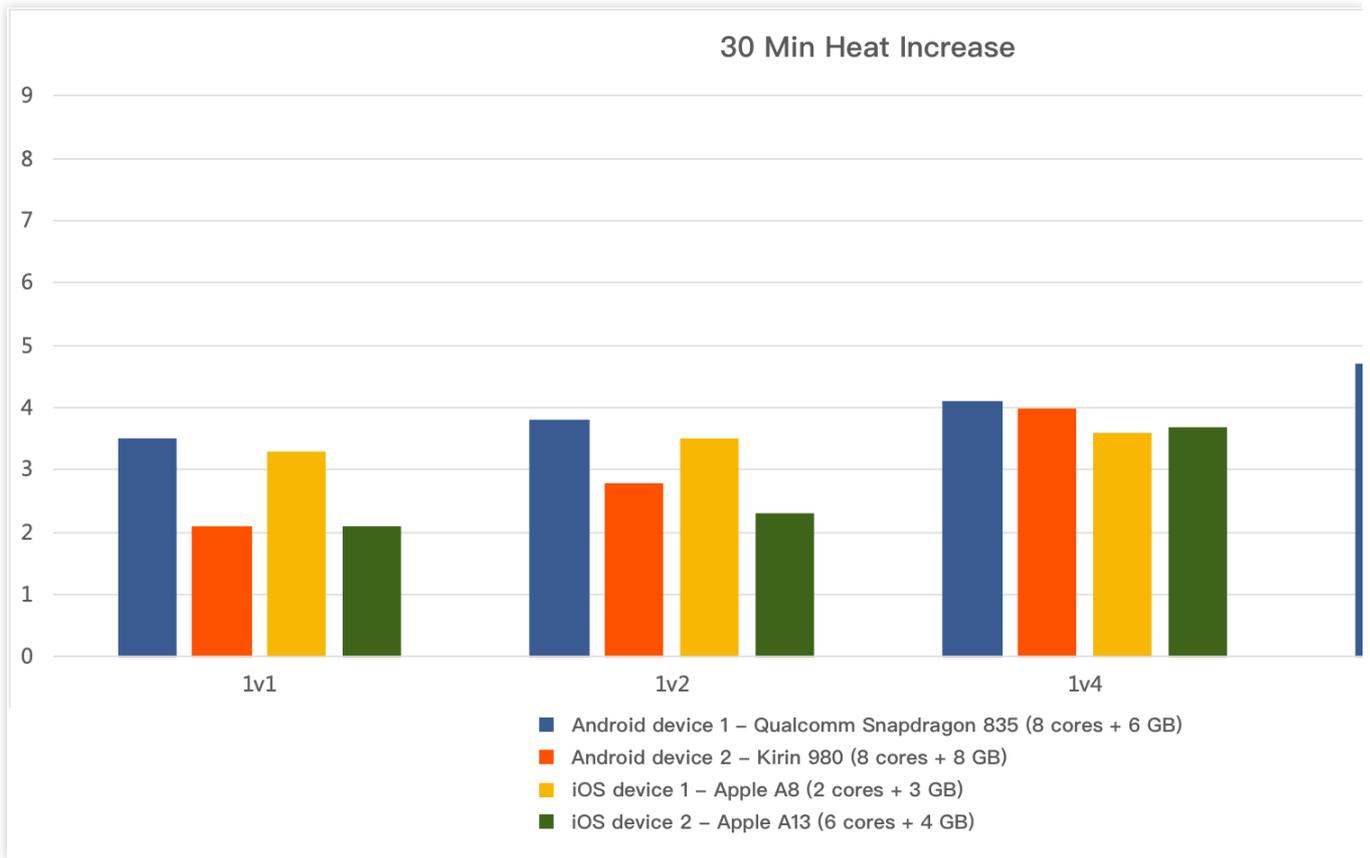
System memory usage



Battery drain after 30 min:



Heat increase after 30 min:



Appendix 1: Network Metrics

| Metric | Description | Example |
|--------|------------------|--|
| Loss | Packet loss rate | 50%: for every 10 packets sent, 5 are lost. |
| Delay | Network delay | 200: Data packets are delivered by the network 200 ms after they are sent by the SDK. |
| Jitter | Network jitter | 300: Packet sending may be delayed 20 ms, 50 ms, 250 ms, 280 ms, or any value up to 300 ms. The average delay is 150 ms. |

Appendix 2: Performance Metrics Under Poor Network Conditions

| Performance Metric | Description |
|--------------------|---|
| MOS | An important measure of the audio quality of telecommunication systems. MOS is generated by |

| | |
|-----------------------------|---|
| | Spirent Nomad using the POLQA standard. The higher the score, the higher the audio quality. |
| End-to-end latency | The time from when audio is captured at the sender end to when it is played back at the recipient end |
| Poor network tolerance test | Spirent Nomad is used to score the SDK under different poor network conditions using the POLQA standard. Foreman video sequences are used to send data, and frame intervals are monitored at the recipient end. Data is collected at 30 points over a course of 10 min or longer. If there are perceptible abnormalities of 3 min at more than 3 data points, or the SDK is unavailable for a relatively long period of time, the SDK is considered intolerant of the network conditions. |

notice

Perceptual Objective Listening Quality Analysis (POLQA) is the ITU-T P.863 standard. It is a globally applicable standard used to score speech quality under different network conditions.

Appendix 3: SDK Performance Indicators

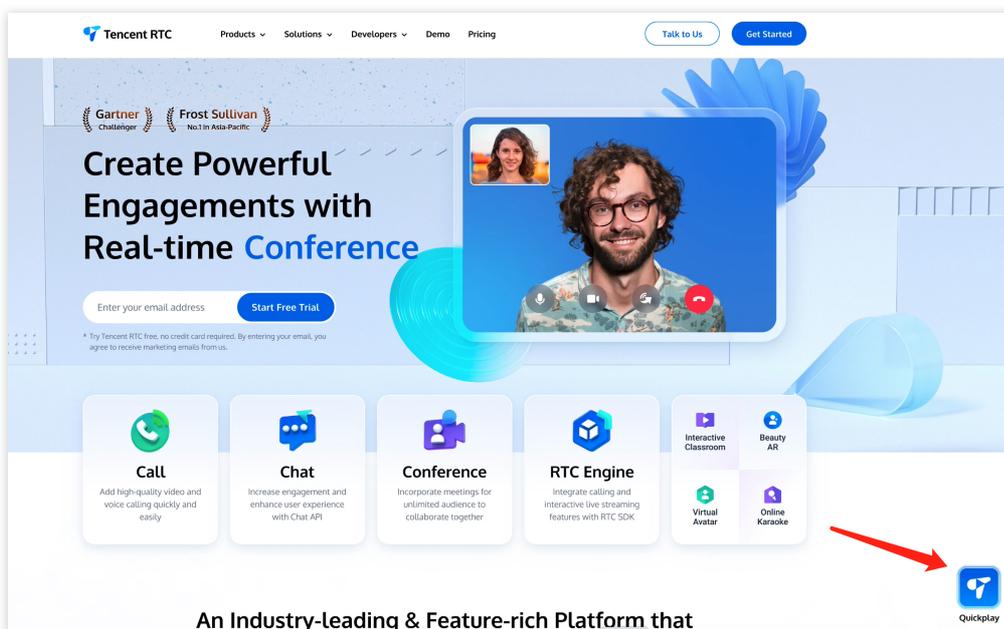
| Indicator | Description | |
|------------------|---|---|
| App CPU usage | Android | Non-normalized CPU usage of the app, which is the same as the results generated by Android Studio Profiler |
| | iOS | CPU usage of the app, which is the same as the results generated by XcodePerfDog usage = Xcode usage / Number of cores |
| System CPU usage | Android | Non-normalized CPU usage of the device, which is the same as the results generated by Android Studio Profiler |
| | iOS | CPU usage of the device, which is the same as the results generated by XcodePerfDog usage = Xcode usage / number of cores |
| Memory usage | Android | Proportional set size (PSS), which is the same as the results generated by Android Java API and Meminfo |
| | iOS | Xcode memory, which is obtained via debug gauges |
| Battery drain | Decrease in battery percentage after 30 min (calculation starts the moment the battery percentage drops from 100% to 99%.) | |
| Heat increase | Temperature is measured with a thermometer when the app is not launched. Then run the app for 30 min under different scenarios. Heat increase = Temperature after 30 min – Temperature when the app is not launched | |

Tencent RTC Quickplay: Experience Ultimate Real-Time Audio and Video Interaction!

Last updated : 2024-01-25 16:42:21

What is Tencent RTC Quickplay?

RTC Quickplay is a web-based real-time video call demo developed using the RTC Engine SDK and Beauty AR SDK. No registration is required - simply click "**Create Room**" to start experiencing audio and video calls instantly.

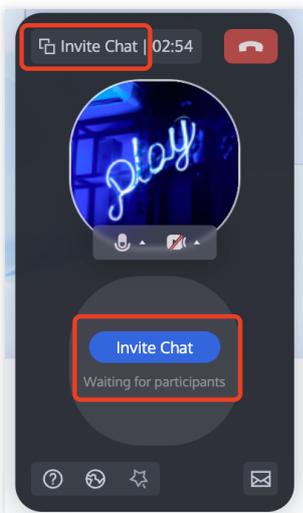


Features and Usage

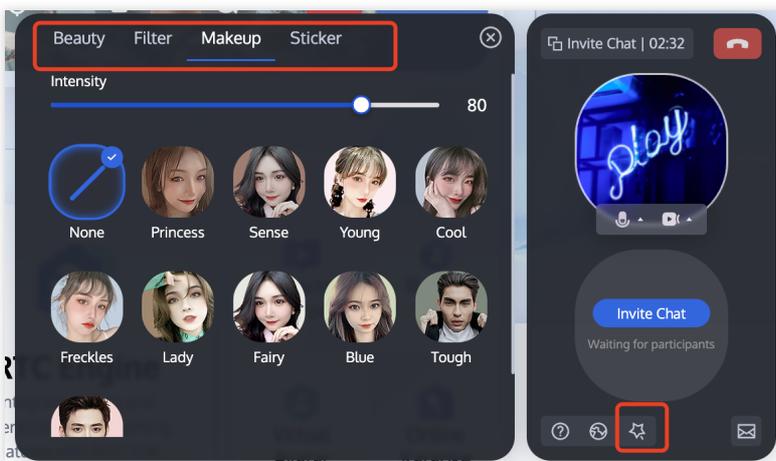
1. **Create Room:** Start your own audio and video call room. (Please confirm that you have enabled the microphone and camera permissions in your system settings or browser settings.)



2. **Invite Friends:** Share your room link to invite others.



3. **Beauty AR:** Enhance your appearance during video calls.



4. **Call Duration Limit:** As RTC Quickplay is for experience use only, there is a certain limit on the experience duration. If you are not logged into Tencent RTC, your single call duration is 3 minutes. If you are logged into Tencent

RTC, your single call duration is 15 minutes. If you exit the room or exceed the call duration, you can recreate a room to start a new call.

How to develop a "RTC Quickplay-like" application?

As a professional real-time audio and video solution provider, Tencent RTC offers a variety of SDK products to meet your diverse business needs.

Call SDK: If you want to develop an application similar to RTC Quickplay, the **Call SDK** is the most suitable and cost-effective solution. It includes 1-to-1 Call, group call, video call, voice call, and more. It also provides **UIKit** and supports multiple platforms including web, iOS, Android, and Flutter, which can accelerate the time-to-market of your product.

Conference: A low-code meeting SDK with UI, helping you integrate a "Zoom-like" feature in your application.

RTC Engine: A customizable RTC SDK with powerful features, enabling you to quickly build real-time audio and video call functions in any scenario.

Beauty AR: Includes beauty mode, filters, stickers, and other AI effects. All in one SDK to make your application more attractive.

Tencent RTC pricing

All Tencent RTC accounts activated can receive free 10,000 minutes per month. The free minutes can be used to deduct minutes generated by Audio and Video Calls, Interactive Live Streaming, On-cloud Recording, and On-Cloud MixTranscoding.

More billing details, please see [billing overview](#).

If you are interested in building an audio and video application, or would like to request a demonstration, please feel free to [contact us](#).