Tencent Real-Time Communication Product Introduction

Tencent Cloud

Contact Us

We are committed to providing personalized pre-sales consultation and technical after-sale support. Don't hesitate to contact us at 4009100100 or 95716 for any inquiries or concerns.

Contents

Product Introduction
- Overview
- Basic Concepts
- Product Features
- Advantages
- Use Cases
- Performance Data

Product Introduction
Overview

Last updated：2025-03-25 10:30:27

Tencent Real-Time Communication (TRTC), leveraging Tencent's deep accumulation in network and audio/video technology over the years, offers two scenario-based solutions: multi-person audio and video calls and low-latency interactive live streaming. Through Tencent Cloud services, it is made accessible to developers, aiming to help them quickly build low-cost, low-latency, high-quality audio/video interactive solutions.

- Multi-person audio and video calling solution
  Relying on Tencent Cloud's global dedicated network, global interconnectivity is achieved. It provides client SDKs and cloud-based APIs covering mobile phones and desktops across the platform. End users can also use the TRTC service in WeChat, QQ, and WeCom Mini Programs, and it can be easily used on webpages.
- Low-latency interactive live streaming solution
  Based on industry-leading network and audio/video technology, combined with Tencent Cloud's high-quality node resources, it helps developers build interactive live streams with lower stutter rate and delay within 1 second, bringing live streaming into the CDN 2.0 era.

The following video will help you quickly understand TRTC.
Watch video

Product Architecture

Tencent Real-Time Communication (TRTC) focuses on cross-platform interoperability for multi-person audio and video calls and low-latency interactive live streaming solutions. It provides SDKs for platforms such as mini programs, Web, Android, iOS, HarmonyOS, Electron, Windows, and macOS, making it easy for developers to quickly integrate and connect with the TRTC cloud service backend. Through mutual integration of different Tencent Cloud products, TRTC can also be quickly and easily used in collaboration with other Tencent Cloud services such as Instant Messaging (IM), Cloud Streaming Services (CSS), and Video on Demand (VOD), expanding into more business scenarios. The product architecture is shown in the figure below:

Platform Support

Tencent Real-Time Communication (TRTC) is **the solution that truly achieves cross-platform interoperability in the industry.** The specific platform support and development environment requirements are listed in the table below:

> **Note:**

HarmonyOS SDK has been released. This version has completed basic function verification. Currently, trial applications are preferentially open to some enterprise-level paying customers. If you need to use it, please fill in the form to contact your business.

Platform	Development Environment Requirement
iOS	• Support iPhone or iPad devices running iOS 9.0 and above versions • Xcode 9.0+ • The project has been configured with an effective developer signature.
Android	• Android Studio 3.5+ • Recommend using Android 4.1 (SDK API Level 16) or higher system.
Windows	• Support Windows 7 and above versions • Visual Studio 2010 and above versions, recommended for use Visual Studio 2015 • .Net Framework 4.0 and above versions
Mac OS	• Xcode 9.0+ • Mac physical device with OS X10.10+ • The project has been configured with an effective developer signature.
Web	• Recommended for use: Desktop client Chrome 56+. Please refer to Quick Integration (Web) for detailed development environment requirements.
Electron	• Support Windows 7 and above versions, Mac OS 10.10 and above versions. • Support Electron 4.0.0 and later versions. Recommended for use: the latest version of Electron SDK.
WeChat Mini Program	• Minimum version requirement of WeChat App for iOS: 7.0.9 • Minimum version requirement of WeChat App for Android: 7.0.8 • Minimum version requirement of mini program base library: 2.10.0 • Since the mini program development tool does not support native components (i.e., <live-pusher> and <live-player> tags), it needs to be run on a real device for experience.
Flutter	iOS: • Support iPhone or iPad devices running iOS 9.0 and above versions • Xcode 9.0+ • The project has been configured with an effective developer signature.

	Android: • Android Studio 3.5+ • Recommend using Android 4.1 (SDK API Level 16) or higher system.
uni-app	• Recommend using the latest HBuilderX Editor. • iOS 9.0 or later devices that support audio and video. Emulator not supported. • Android devices with Android version not less than 4.1 and support for audio and video. Emulator not supported. If it is a real device, enable the allow debugging option. • iOS/Android devices have already connected to the Internet.

Basic Concepts

Last updated: 2025-03-25 10:32:05

This document aims to introduce some basic concepts that you may encounter in the service process of using tencent real-time communication (TRTC).

Application

TRTC manages different businesses or projects in the form of applications. You can create different applications for different businesses or projects respectively in the TRTC console, thereby realizing the isolation of business or project data. Each cloud account can create up to 100 TRTC applications.

SDKAppID

SDKAppID (application identifier/app ID) is a unique identifier used by Tencent Cloud Backend to distinguish different TRTC applications. It is automatically generated when creating an application in TRTC Console. The data of different SDKAppIDs are not interconnected.

UserID

UserID (user identity) is used to uniquely identify a user in a TRTC application and is specified by the developer.

- User identity is the mapping of the account that a user logs in to the developer's business system on Tencent Cloud. Under normal circumstances, developers can directly use the username as the UserID.
- The length of the parameter value range should not exceed 32 bytes. Please use English characters, digits or underscores. It cannot be all digits. Case-sensitive.

Room

A room is an audio and video space. Users in the same room can receive each other's real-time audio and video data.

- TRTC uses the virtual concept of a room for mutual isolation between users.
- Only users in the same room can receive audio and video from each other.
- The same UserID can only be in one room at the same time. If there is another UserID entering the same room, the previous UserID will be removed from the room.

Room Lifecycle

- The first user to enter the room is the owner of the current room, but this user is unable to proactively dismiss the room.

- **In call mode**: When all users voluntarily check out, the room will be dissolved immediately in the backend.
- **In live streaming mode**: When the last checkout user is in the anchor role, the room will be dissolved immediately in the backend; when the last checkout user is in the audience role, the backend will wait for 10 minutes before dissolving the room.
- If a single user in the room has an unexpected disconnection, the server will remove this user from the current room after 90 seconds. If all users in the room have an unexpected disconnection, the server will automatically dissolve the current room after 90 seconds. **The waiting duration for user abnormal disconnection will be incorporated into billing duration statistics.**
- When a user attempts to join a room that does not exist, the TRTC backend automatically creates a room.

RoomID

RoomID is used to uniquely identify a room in a TRTC application. RoomID is divided into digit type (roomId) and string type (strRoomId). Note that strRoomId and roomId cannot be used together. "123" and 123 are not considered the same room in the TRTC backend service.

UserSig

UserSig is a security protection signature designed by Tencent Cloud for performing login authentication on a user, confirming the authenticity of the user, and preventing malicious attackers from misappropriating your cloud service usage rights. For details, see the UserSig Related Issues document.

Push

Push refers to the operation where a user uploads local audio and video data to the TRTC server, corresponding to "streaming".

Subscription

Subscription refers to the operation where a user requests to pull the audio and video data of designated users from the TRTC server, corresponding to "stream pulling".

Role

TRTC supports two types of roles: **anchor** (TRTCRoleAnchor) and **audience** (TRTCRoleAudience). The difference between them is:

- The anchor role can not only push its own audio and video data to the server, but also subscribe to and play the audio and video data of other anchor roles from the server.

- Audience role **only supports** subscribing to and playing the audio and video data of the anchor role from the server.

In call mode, all users entering the room are in the anchor role. In live streaming mode, based on actual business scenes, you can divide users entering the room into two types of roles: anchors and audience. The same user can switch roles at any time.

CDN Live Streaming Viewing

CDN Live Streaming Viewing, also known as "CDN bypass live streaming". TRTC uses the bypass transcoding cluster in the cloud to convert the UDP protocol used by TRTC into the standard live streaming RTMP protocol, push TRTC audio and video data to the standard Cloud Streaming Services system, and then distribute it via CDN, thereby achieving CDN Live Streaming Viewing. For details, see Implementing CDN Live Streaming Viewing document.

Cloud Recording

TRTC uses the bypass stream pushing method to utilize the capabilities of Cloud Streaming Services to offer you full-process on-cloud recording features (i.e., audio recording/video recording), and stores the recorded files on the Video on Demand platform, ensuring the reliability and real-timeness of the recording process. For details, see On-Cloud Recording and Playback document.

Cloud-Based Mixed-Stream Transcoding

In application scenarios such as **CDN Live Streaming Viewing** and **Cloud Recording**, you may need to mix multiple audio and video streams in a TRTC room into one. You can use the MCU hybrid transcoding cluster in the TRTC cloud service backend to complete this work. The MCU cluster can mix multiple audio and video streams as needed and distribute the final generated video stream to live stream CDN and cloud recording system. For details, see Cloud-Based Mixed-Stream Transcoding document.

Dumb Terminal

When a dumb terminal enters a room as an audience to pull a stream, it will not be perceived by other SDKs (the far end fails to receive the room entry and exit event notifications of the dumb terminal).

Product Features

Last updated: 2025-03-25 10:45:14

Basic Audio and Video Features

Feature	Feature Description	Common Application Scenarios	Description
Video Call (VideoCall)	• Two-person or multi-person video call, supporting HD quality of 720P, 1080P, 2K and 2K+ (specific device). • The number of people in a single room is limited to 300, and a maximum of 50 people can turn on the camera at the same time.	1-on-1 video call, 300-person video conference, online consultation, video chat, video customer service, video interview, video dual-recording, online insurance claim settlement, video-based party games, etc.	• Audio and Video Duration Billing • prepaid package
Voice Call (AudioCall)	• Two-person or multi-person audio call, supporting 48kHz and stereo. • The number of people in a single room is limited to 300, and a maximum of 50 people can turn on the microphone at the same time.	1-on-1 voice call, group audio call, voice chat, audio conferencing, audio customer service, Werewolf, etc.	

Interactive Live Streaming (LIVE)	• Support video communication interaction between anchors and audiences. • Support anchors to PK across rooms (live streaming rooms). • Support smooth microphone switching. No waiting during the switching process. Anchor latency is less than 300 ms. • The number of people in a single room is unlimited. A maximum of 300 anchor roles can be set. A maximum of 50 people can turn on the camera at the same time. • Support 100,000 audiences to play simultaneously in low-latency live streaming mode. Playback latency is as low as 1000 ms. • Under CDN bypass live streaming mode, the number of audiences is unlimited.	Low-Latency Video Streaming, Interactive classrooms for up to 100,000 participants, Live streaming PK, Video Blind Date Room, Interactive course, Remote training, Large conference	
Interactive Voice Live Streaming (VoiceChat Room)	• Support voice mic-connection interaction between anchors and audiences. • Support anchors to PK across rooms (live streaming rooms). • Support smooth microphone switching. No waiting during the switching process. Anchor latency is less than 300 ms. • The number of people in a single room is unlimited. A maximum of 300 anchor roles can be set. A maximum of 50 people can turn on the microphone at the same time.	Low-latency audio streaming, live audio co-anchoring, audio live streaming PK, voice chat room, voice dating room, Karaoke room, FM radio, etc.	

	- Support 100,000 audiences to play simultaneously in low-latency live streaming mode. Playback latency is as low as 1000 ms. - Under CDN bypass live streaming mode, the number of audiences is unlimited.		
On-Cloud Recording	Use the MCU recording cluster to record the upstream audio and video streams in TRTC rooms as single streams or merged streams (mixed streams) as needed, and store the recorded files on the Video on Demand platform, ensuring the reliability and real-time ness of the recording process.	Dual recording, archiving, compliance, etc.	On-cloud recording belongs to value-added service, and requires an extra fee for on-cloud recording fee.
Cloud-based Mixed-stream Transcoding	Use the MCU cluster to perform mixed-stream transcoding on the upstream audio and video streams in TRTC rooms as needed. The transcoded audio and video streams can be bypass streamed to Cloud Streaming Services for cloud recording or to realize CDN live streaming viewing.	Multi-screen display mixed as needed, recording format conversion, etc.	Cloud-based Mixed-stream Transcoding belongs to value-added service, and requires an extra fee for cloud-based mixed-stream transcoding fee.
CDN Live Streaming Viewing	Also known as "CDN bypass live streaming". TRTC uses a bypass transcoding cluster in the cloud to convert the UDP protocol used by TRTC into the standard live streaming RTMP protocol, pushes TRTC audio and video	Interactive live streaming, live sharing, large conference, remote live audience viewing	Relayed live streaming belongs to value-added service. Relevant

	data to a standard CSS system, and then distributes it via CDN, thereby achieving CDN live streaming viewing.		fees are collected by CSS. For details, see CDN Live Streaming Viewing > Relevant Fees.

Value-Added Audio and Video Features

Feature	Feature Description	Common Application Scenarios	Description
AI noise reduction	AI noise reduction can eliminate sounds that traditional noise reduction cannot, such as coughing, sneezing, car honks and other non-stationary noises. In IM scenarios, it supports real-time audio noise reduction calls and AI noise reduction for locally recorded audio messages. Initiation method: In startLocalAudio, select **TRTCAudioQualitySpeech** for audio quality.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Online Class, etc.	Unlock value-added audio and video features by subscribing to the TRTC monthly subscription package.
Weak network call lag optimization	Optimize the lag rate in outdoor and other weak network environments, with a faster instant opening speed.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Online Class, etc.	
Mini Program Call Acceleration	Tencent Real-Time Communication (TRTC) has comprehensively optimized and upgraded the interconnection between RTMP over Quic and TRTC, enhancing the call smoothness and stability of the Mini Program SDK.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Online Class, etc.	

input media stream into room	TRTC supports streaming into room via the RTMP standard protocol as well as pushing online media stream into room, and flexible selection can be made according to the user scenario.	Online classes, sports event viewing together, etc.	
Page recording	Page recording provides the ability to record any web page in the cloud and store it, realizing on-demand playback anytime and anywhere.	Online classes, etc.	
3D Stereo Sound Effect	Enable users to have spatial sense in sound experience, and present different audio effects according to the face orientation of virtual characters, orientation of audio sources, orientation and location, and height, to perfectly simulate life-like sound experience.	Voice chat room, Karaoke room, FM radio, virtual concert, online games, etc.	
Voice changing effect	Voice changing effects can act on the voice and perform secondary processing on the voice through acoustic algorithms to obtain a timbre different from the original sound.	Voice chat room, Karaoke room, FM radio, virtual concert, online games, etc.	
Layered video encoding	Combined with the O264RT encoding technology launched by Tencent Multimedia Lab, enhance the image loading speed, obviously reduce bandwidth consumption, and make terminal adaptation more stable. **Note**: After purchasing the Monthly Package for the first time, go to Console > Application Management > Feature Configuration > Value-added Feature and enable this	Video Call, Online Class, Interactive Live Streaming, etc.	

	capacity to take effect automatically.		
Area of Interest Video Encoding	Area of Interest Video (ROI) encoding can lower CPU consumption and ensure that in most scenarios, it can provide a video experience with the "lowest latency" and "optimal quality" for the current scenario.	Video Call, Online Class, Interactive Live Streaming, etc.	
High-resolution quality	For demand scenes of high-resolution quality, support streaming into room and screen-sharing feature in 2K/4K resolution (currently only the PC and Web support 2K+ resolution).	Video Call, Online Class, Interactive Live Streaming, etc.	
SDK Private Encryption	Support users to enable secondary custom encryption, which can be transmitted securely according to the encryption and decryption algorithm you specify. Note: After purchasing the TRTC Monthly Package - Flagship Edition, you need to submit an application and it will be available to use this feature after review passed (compliance requirements).	High-level encryption scenarios such as finance	
Real-Time Online Room and User Query API	Provide Rest API for online room and user query, applicable to active data statistics, not recommended for strong business logic judgment.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Online Class, etc.	
Virtual Background	• Based on portrait contour recognition technology, achieve accurate portrait segmentation effect, support background blur and background replacement.	Video Call, Interactive Live Streaming, Online Meeting, Online Classes, etc.	

	• **Currently**, Mobile Client, Web **is available**. More features and platform support are coming soon.		
Video Screenshot Upload	Support through the SDK API to initiate the screenshot upload feature. Users can use screenshots for third-party audits, cover image settings, and other scenarios.	Video Call, Online Class, Interactive Live Streaming, etc.	

More Advanced Functions

Feature	Feature Description	Common Application Scenarios	Description
Interactive Co-anchoring	Support mic-connection interaction. The audience can freely and smoothly switch the microphone. No waiting during the switching process.	Interactive Live Streaming, Online Class, Chat Room, etc.	Using the interactive mic functionality will generate basic service usage and require payment of audio and video duration fee.
Cross-room PK	Also known as "cross-live streaming room PK", multiple anchors PK interactively across rooms while audiences view.	Showroom streaming, PK mic connection, cross-room teaching, etc.	Using the cross-room PK feature will generate basic service usage and require payment of audio and

			video duration fee.
Monitoring dashboard	**Room call survey**: A dashboard that records the call quality details of each room, allowing developers to view call details and information through the monitoring dashboard and understand the call status of end users. **Real-time monitoring:** Real-time quality supports real-time viewing of scale and quality metrics under the room, provides multiple metrics, and offers multi-dimensional analysis. **Data dashboard:** Provides analysis features for viewing scale data and quality data of business operations on the application dimension. It can help you quickly understand the call scale data and trends under the application (sdkappid). **Monitoring and alarming:** Provides proactive alarm features and supports setting alarms for custom call metrics. When an alarm is triggered, you can receive alarm notifications by mail, API callback, and other methods, and take timely countermeasures.	ALL scenarios	Provides free version basic features. If you need to use advanced functions, you need to activate the Monitoring dashboard package service.
Screen Sharing	Also known as screen sharing, it supports sharing local computer desktops, windows, and screen areas with others, such as the	Online classes, slide sharing, remote assistance, etc.	Using the screen-sharing feature will generate

	window for playing PPTs in Microsoft PowerPoint.		basic service usage and require payment of audio and video duration fee.
High-quality audio	Support high-quality audio with 48kHz sampling, full-link 128kbps high-quality, true left and right channel stereo audio, to achieve clear listening experience for room users and immersive interactive experience.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, High-audio-quality FM, Music Teaching Class, Karaoke Room, Online Classes, etc.	Free
3A handling	Industry-leading Tencent Ethereal Audio Lab provides 3A processing algorithms, delivering better audio quality in scenarios like dual-talk and noise reduction. 3A refers to AEC (acoustic echo cancellation), ANS (automatic noise suppression), and AGC (automatic gain control).	ALL audio scenarios	Free
Basic beauty filter	Support basic beauty filter capabilities, including set whitening, skin smoothing, rosy as well as basic filter effects.	Video Call, Interactive Live Streaming, Online Classes, etc.	Free
BGM	Support adding local music files in MP3, AAC, WAV and other formats as the background music for voice.	Voice Call, Video Call, Interactive Live Streaming, Online Classes, Voice Chat Room, Karaoke Room, FM Radio, etc.	Free

Audio effects	Add audio effects during call, such as applause, cheering, whistling, booing, etc.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Karaoke Room, FM Radio, etc.	Free
Accompaniment	Send the locally played sound, such as the sound played by QQ Music Player on a computer, to others.	Interactive Live Streaming, Online Classes, Voice Chat Room, FM Radio	Free
Reverberation	Provide reverb effects, such as those in KTV, small rooms, concert halls, bathrooms, etc.	Voice Call, Video Call, Interactive Live Streaming, Voice Chat Room, Karaoke Room, FM Radio, etc.	Free
Volume level callback.	Provide the volume value for easy display as waveform animation or prompt.	Voice Call, Video Call, Voice Chat Room, FM Radio, Karaoke Room, Voice Detection	Free
IEMs (In - Ear Monitors)	Play the locally recorded sound through local headphones so that oneself can hear the sound emitted by oneself. It is generally used for detecting speech error or pitch control.	Interactive Live Streaming, Showroom Streaming, Karaoke Room, etc.	Free
Custom Audio Data	Support self audio capture callback. Developers can process original data and perform customized operations, such as external non-standard devices, audio files, etc.	Non-standard device integration, custom audio effect, speech processing, speech recognition, etc.	Free
Custom Video Data	Support custom video sources and renderers. Use non-camera video sources, such as video files, external devices, and third-party data sources.	Custom beauty filters, custom data source, multi-device management,	Free

		video recognition, image process	
SEI Information	Embed custom information into the video stream through SEI frames and synchronize it to other users, such as lyrics, titles, etc.	Karaoke Room, Live Quiz Broadcasting, Interactive Live Streaming, etc.	Free

Extended Features

> **Note:**
> Extended features are value-added services provided by Tencent's TRTC products combined with other Tencent Cloud products. Relevant fees will be charged separately by other cloud products according to their respective billing rules.

Feature	Feature Description	Common Application Scenarios	Description
Chat	Chat messages, comments, bullet screens, gift sending, likes and other features can be implemented through IM one-on-one chat, group chat and chat rooms with no upper limit on the number of people. Signaling interaction can be performed through IM to implement call initiation, room user count statistics and other features.	Online customer service, Interactive live streaming, Interactive classes, Remote training, etc.	Chat belongs to value-added service. Relevant fees are charged by Chat. For details, see Chat relevant fees.
Tencent Interactive Whiteboard	Multi-user online real-time shared whiteboard interaction can synchronize the whiteboard pen strokes drawn by the user to other users.	Interactive classes, Remote training, Remote sharing, etc.	The interactive whiteboard belongs to value-added services. Relevant fees are charged by Interactive Whiteboard. For details, see

			Interactive Whiteboard Purchase Guide.
Beauty Effect	Provide value-added capabilities such as intelligent beauty feature, special effects filter, makeup sticker, virtual background and gesture recognition.	Video call, Interactive live streaming, Showroom streaming	AI beautification belongs to value-added services. Relevant fees are charged by Tencent Effect SDK. For details, see Tencent Effect SDK Snapshot Pricing.
Voice content review	Support violation content security detection such as porn information detection in speech, applicable to business content security inspection.	Business security inspection, compliance, etc.	Voice content review is a value-added service. Tencent Cloud Content Safety charges fees. If you need to use it, please see integration documentation.
Video content review	Support violation content security detection such as video porn detection, applicable to business content security inspection.	Business security inspection, compliance, etc.	Video content review is a value-added service. Tencent Cloud Content Safety charges fees. If you need to use it, please see integration documentation.

Advantages

Last updated: 2025-03-25 10:33:14

Cross-Platform Global Interconnectivity

The industry's truly achieved cross-platform interoperability solution, fully adapted to more than 30,000 terminals.. Provide client SDKs and cloud-based APIs that cover mobile phones and desktops across the platform, supporting global interconnectivity on platforms such as iOS, Android, HarmonyOS, Windows, Mac, Web, etc. Terminal users can also use the TRTC service in WeChat, QQ, and WeCom Mini Programs.

Deep Cooperation with Mini Programs

With WeChat Mini Program Engine Deep Cooperation, TRTC, as a provider of WeChat built-in SDK, enables you to obtain an excellent user experience comparable to Native in WeChat mini programs.

Low-Threshold Quick Integration

Only 2 lines of code are required to run the test Demo, and 10 lines of code to complete Universal Capability Access. Quickly build a low-latency, smooth, high-quality Tencent Real-Time Communication (TRTC) product from scratch. For detailed operation guide, please see API-Example and Importing SDK to Project documents.

Contextual UI Components

Provide various contextual UI components such as Video Call, Group Conferencing, Online Live Streaming, and Voice Chat Room to help developers quickly implement features in the simplest way.

Low-Latency

Provide international link network connection channels with global coverage, high connectivity, high reliability, and strong security. Use our independently developed multi-level optimal addressing algorithm and have whole network scheduling capability. Have various high-bandwidth resource reserves and a global node layout. **Ensure that the average end-to-end latency of international links is less than 300 ms.**

Low Stutter

Reduce buffering through intelligent network quality control and coding optimization. Actual test shows that **the packet loss resistance exceeds 80%,and the resistance to network jitter**

exceeds 1000 ms. In a weak network environment, it can still ensure high-quality audio and video communication and ensure a smooth and stable audio/video communication process.

High-Quality

Supports 720P, 1080P, 2K and 2K+ high-definition quality. Video is normal at 70% packet loss rate. In terms of audio, it supports high-quality audio sampled at **48kHz**. The industry-leading Tencent Ethereal Audio Lab provides 3A processing algorithms to eliminate echo and howling. Full-link 128kbps high-quality, true left and right channel stereo audio achieves clear listening experience for room users and immersive interactive experience.

Use Cases

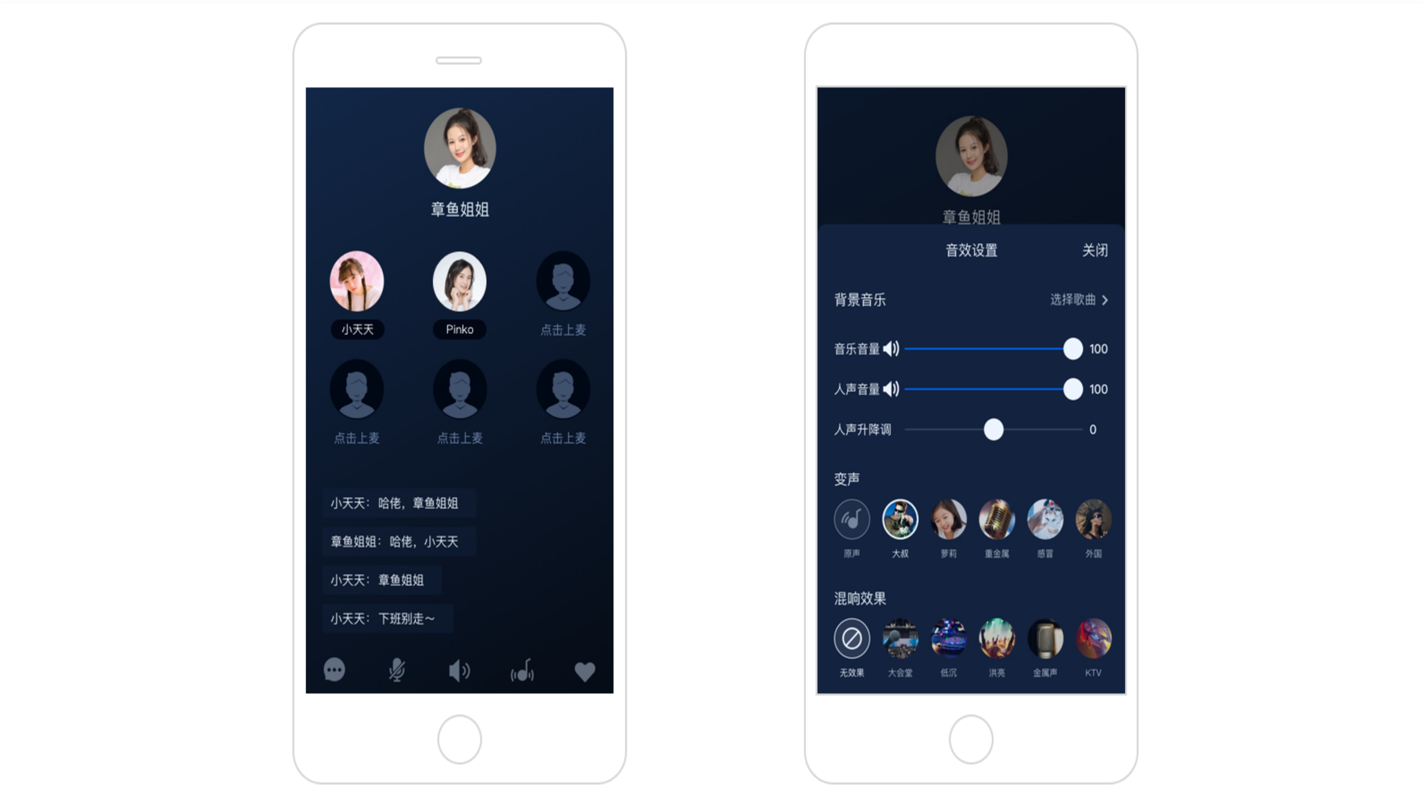
Last updated: 2025-03-25 10:28:46

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

Voice Chat Room

Supports up to 50 people speaking simultaneously, with smooth mic on/off transitions and chat latency of under 300ms; supports voice changing, ambiance sound effects, reverb, and other audio effects to enrich the voice chat experience. Integrates Chat for various interaction forms like public chat, private chat, group chat, liking, and gifting, creating an excellent chat interaction experience. TRTC provides reusable voice chat room components to minimize development costs. For related component usage guidelines, please see Voice Chatroom. Search for the required CAM policy as needed, and click to complete policy association.



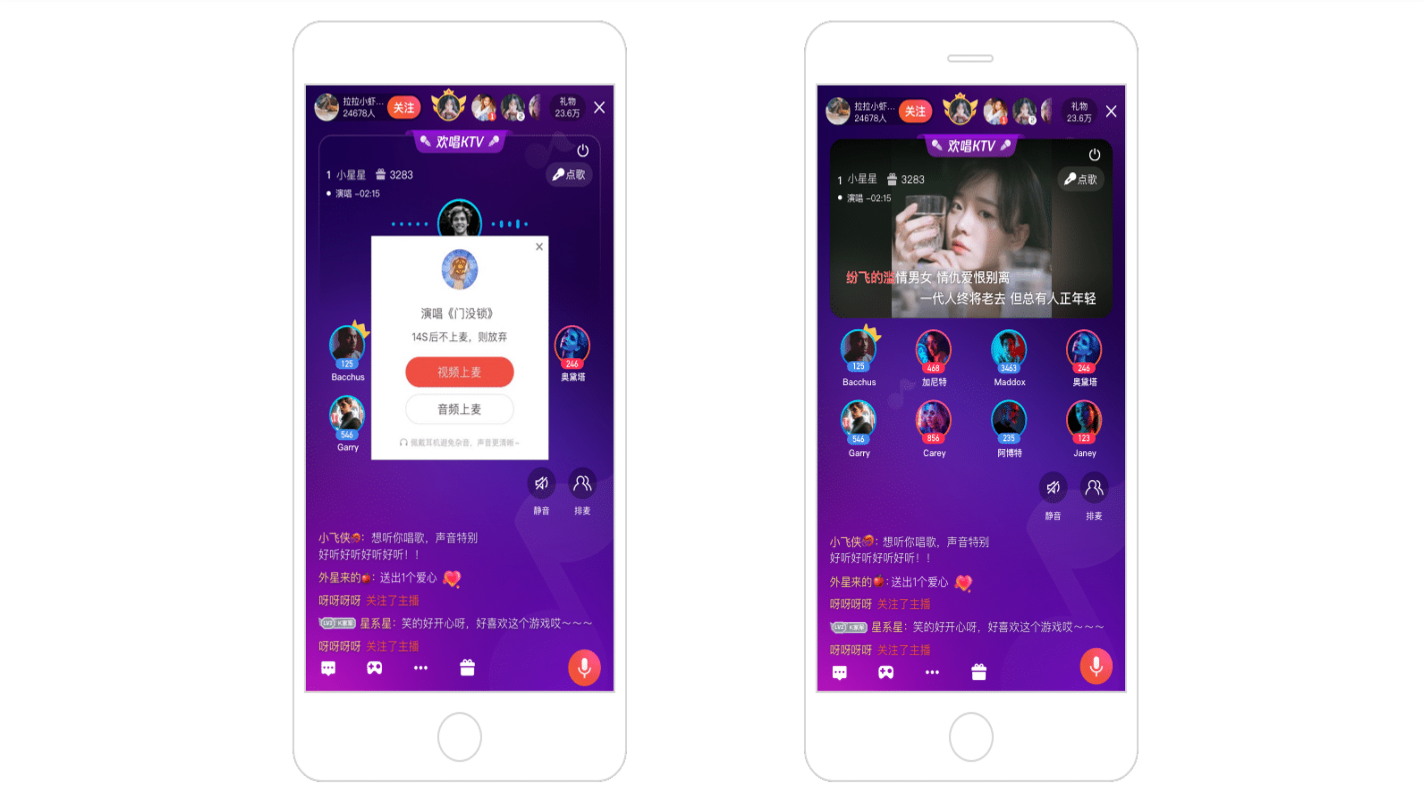
Voice Radio

Supports a maximum sampling rate of 48kHz, a bitrate of 192kbps, and stereo audio; local music in MP3, AAC, WAV, and other formats can be used as background music to easily create a high-quality voice radio. Offers various voice-changing effects like middle-aged man and little girl to make the voice radio more interesting. TRTC provides reusable voice radio components to minimize development costs. For related component usage guidelines, please see Voice Chatroom.
Search for the required CAM policy as needed, and click to complete policy association.

Online Karaoke

Supports a maximum sampling rate of 48kHz, a bitrate of 192kbps, and stereo audio for a studio-quality online singing experience, with duet latency below 300ms ensuring a seamless singing collaboration. Offers various synchronization mechanisms like message pass-through and timestamp to accurately sync accompaniment, vocals, and lyrics in online karaoke scenarios. Supports the ear return feature, allowing you to avoid off-pitch singing.

Search for the required CAM policy as needed, and click to complete policy association.

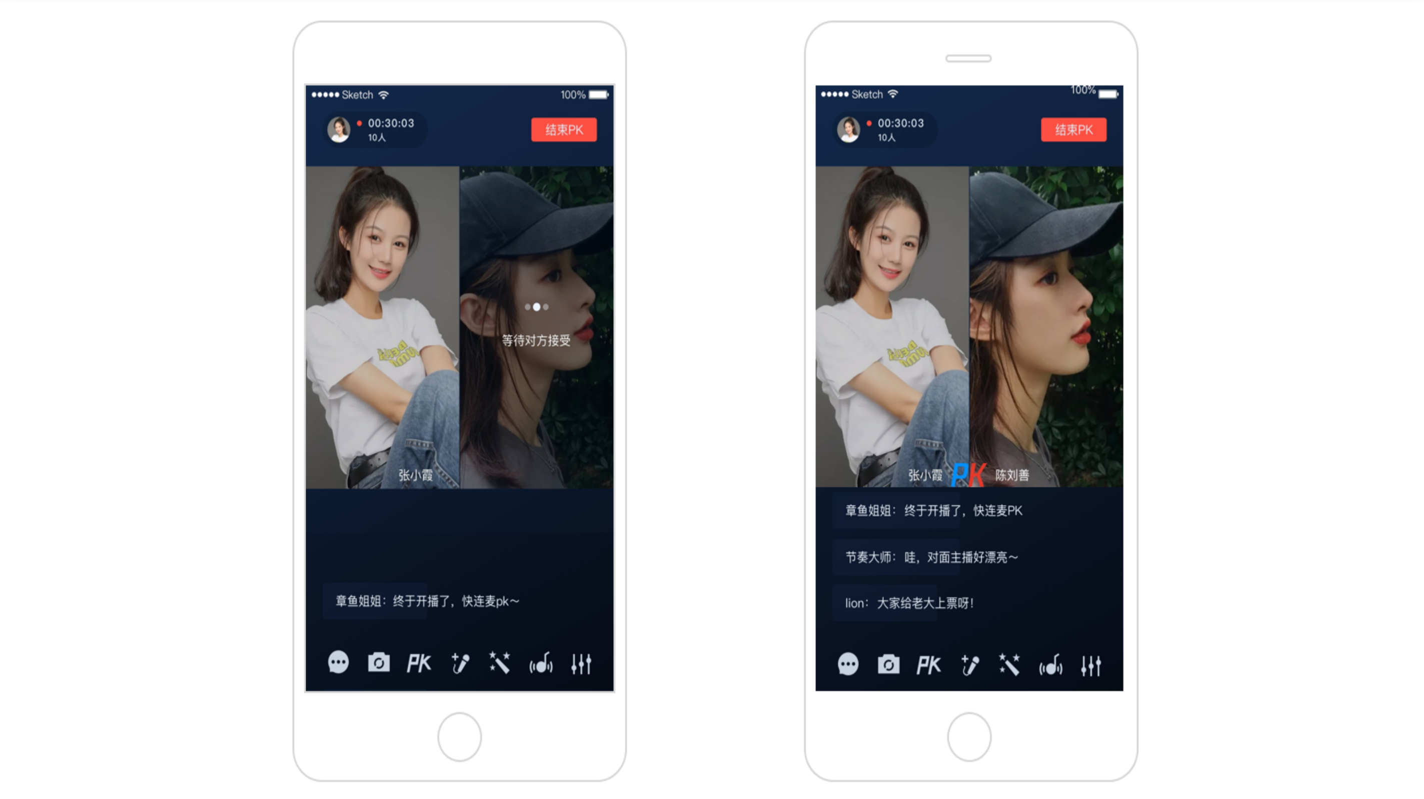


Interactive Video Streaming

Live Show Streaming

Cross-room co-broadcasting PK delay is less than 300ms, supporting audience and host interaction, smooth microphone handover, meeting the high-frequency interaction needs in show live streaming scenarios. Supports intelligent beautification, making show live streaming more charming. TRTC provides scene-based components for show live streaming, which can be directly reused to significantly reduce development costs. For usage guidelines of related components, please refer to Video Interactive Live Streaming.

Search for the required CAM policy as needed, and click to complete policy association.



Interactive big class

Accommodates 100,000 students watching simultaneously with a delay of less than 300ms, supports interactive communication between teachers and students, smooth microphone handover, ensuring uninterrupted classroom interaction. Supports screen sharing, interactive whiteboards, recording and playback among other classroom application features, creating a richer format for online interactive large classes. TRTC offers scene-based components for interactive large classes, which can be directly reused to significantly reduce development costs. For usage guidelines of related components, please refer to Real-time Interactive Classroom.

Search for the required CAM policy as needed, and click to complete policy association.

Interactive small class

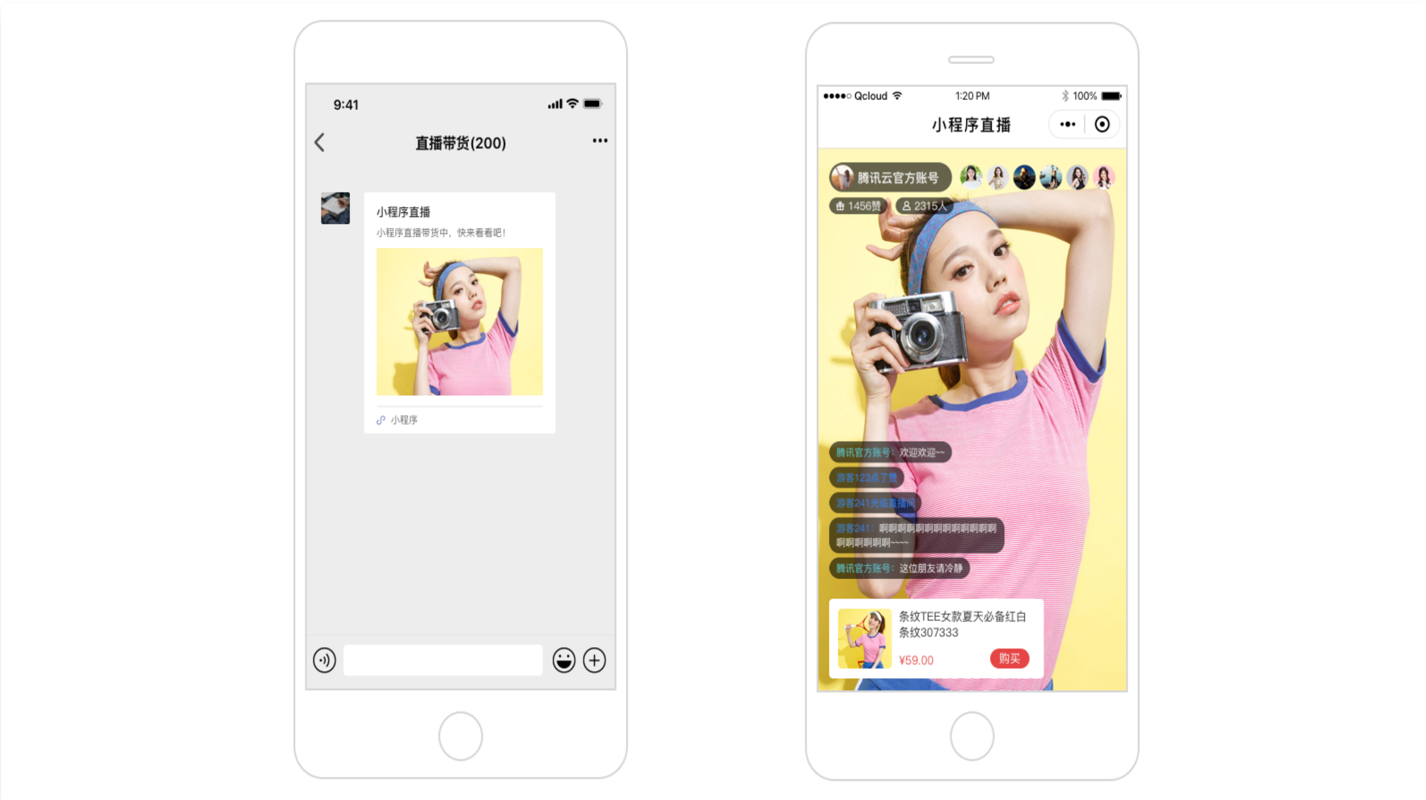
Supports various interactive small class formats like **1v1,1v2,1v6,1v32**, with teacher-student interaction delay less than 300ms, ensuring smoother communication. Supports screen sharing, courseware sharing, interactive whiteboards among other classroom application features, creating a richer online teaching experience. Supports full-class recording, post-class on-demand playback, reinforcing learning outcomes.

Search for the required CAM policy as needed, and click to complete policy association.

Mini Program Live

Mini Programs, APP, and PC platforms are fully interconnected, allowing live content to quickly reach users across multiple platforms. Integrated with instant messaging, VOD, short videos, and other features, it supports live chat bullet screens, likes and gifts, live recording, and playback of recordings to enhance the live streaming experience on Mini Programs. It supports intelligent video content moderation, image and text moderation, instantly handling inappropriate content to ensure compliance and business safety.

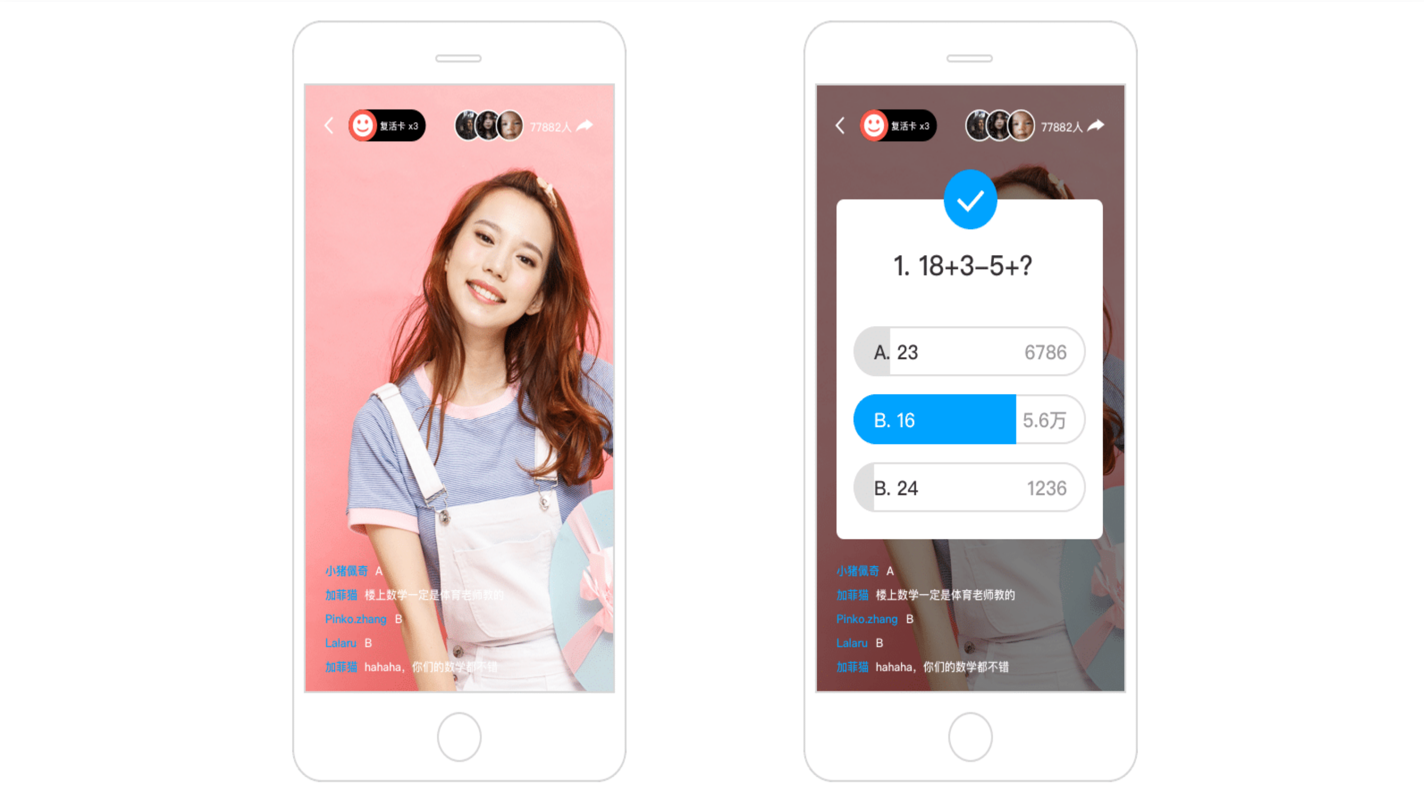
Search for the required CAM policy as needed, and click to complete policy association.



Live Trivia

Supports low-latency online live streaming under high concurrency, minimizing audience viewing delay, suitable for synchronously delivering questions to the audience. Supports message passthrough, timestamps, signaling channels, and various synchronization mechanisms to precisely achieve "synchronized display of audio, video, and questions". Meets the needs for ultra-high concurrency IM interaction, supports interactive competition questions in live streaming, result statistics, and multi-person co-broadcasting interaction, making online quizzes more interesting. Keyword and answer hints are filtered in real time during interactive live streaming to enhance user experience and reduce business compliance risks.

Search for the required CAM policy as needed, and click to complete policy association.



Audio Call

Multi-person Voice Calls

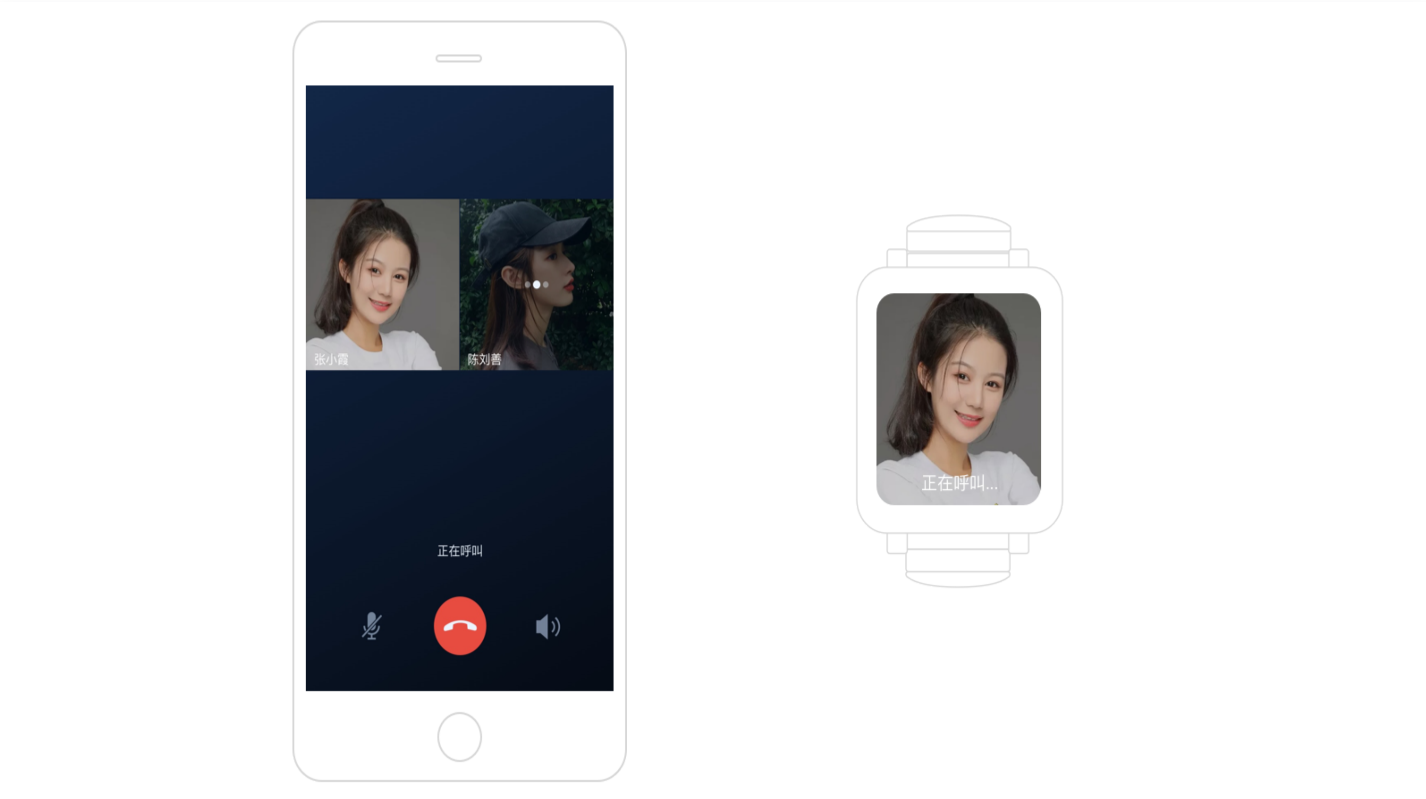
Supports up to 300 people in a call, with a maximum of 50 people turning on their microphones simultaneously, and supports up to 48kHz sample rate and 192kbps bitrate, combined with excellent 3A processing to create a smooth, high-quality voice call experience. TRTC provides scene-based components for multi-person voice calls that can be directly reused to greatly reduce development costs. For usage guidelines of related components, please refer to Real-time Voice Calls.

Search for the required CAM policy as needed, and click to complete policy association.

One-to-one audio call

A latency below 300ms; smooth and stable audio calls under a packet loss rate over 80% and network jitter over 1000ms; ensuring smooth voice communication even in weak network environments. Enriched with Chat, providing a rich set of call signaling management APIs to easily access various voice call scenarios. TRTC offers scenario-based components for one-to-one voice calls that can be directly reused, significantly reducing development costs. For component usage instructions, see Real-time Voice Call.

Search for the required CAM policy as needed, and click to complete policy association.



Werewolf

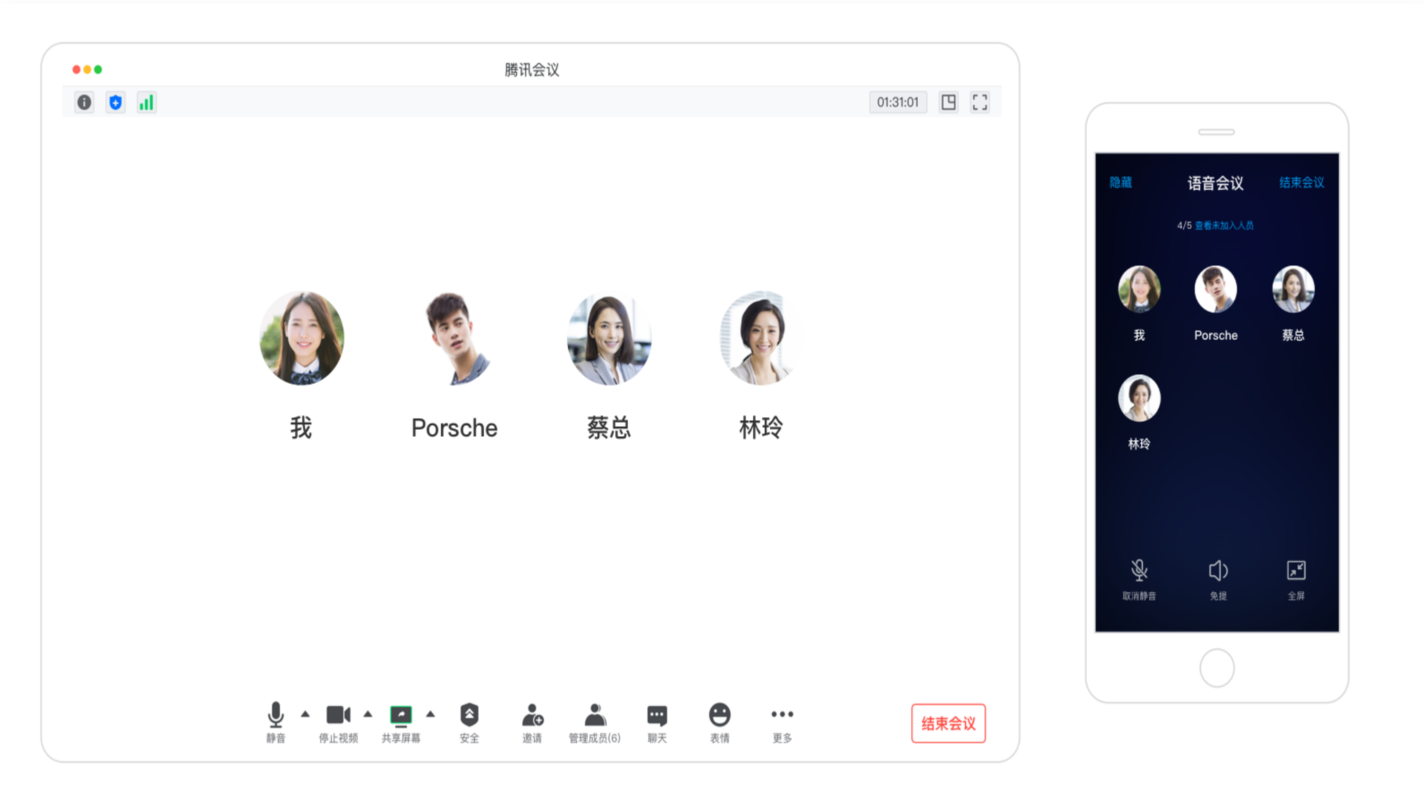
A latency below 300 ms; smooth and stable communication under a packet loss rate over 80% and network jitter over 1,000 ms; real-time network monitoring; audio device testing to ensure that all players can be heard, improving gaming experience.

Search for the required CAM policy as needed, and click to complete policy association.

Audio conferencing

Cross-platform compatibility, allowing flexible joining from mobile phones, computers, tablets, and WeChat; excellent 3A processing ensures echo and howling rejection for smooth and clear conferencing. Supports interactive whiteboard and file sharing, making voice meetings more efficient.

Search for the required CAM policy as needed, and click to complete policy association.



Video Call

Group video call

Supports multi-person video calls with 720p, 1080p, and 2K high-definition quality. Each room allows up to 300 participants, with a maximum of 50 video streams. Supported features include Chat, video on demand, recording, video, and pornography detection for easy access to various applications. TRTC offers scenario-based components for multi-person video calls that can be directly reused, significantly reducing development costs. For component usage instructions, see Real-time Video Call.

Search for the required CAM policy as needed, and click to complete policy association.

One-to-one video call

Support for one-to-one video calls with high-quality video resolutions including 720p, 1080p, 2K, and 2K+ (specific devices). It provides high-quality video call services with the integration of Chat, screen sharing, recording, interactive whiteboard, and other features to meet various application scenarios. TRTC offers a two-person video call scenario component for direct reuse, significantly reducing development costs. For component usage guidelines, please refer to Real-time Video Call.

Search for the required CAM policy as needed, and click to complete policy association.

Online conferencing

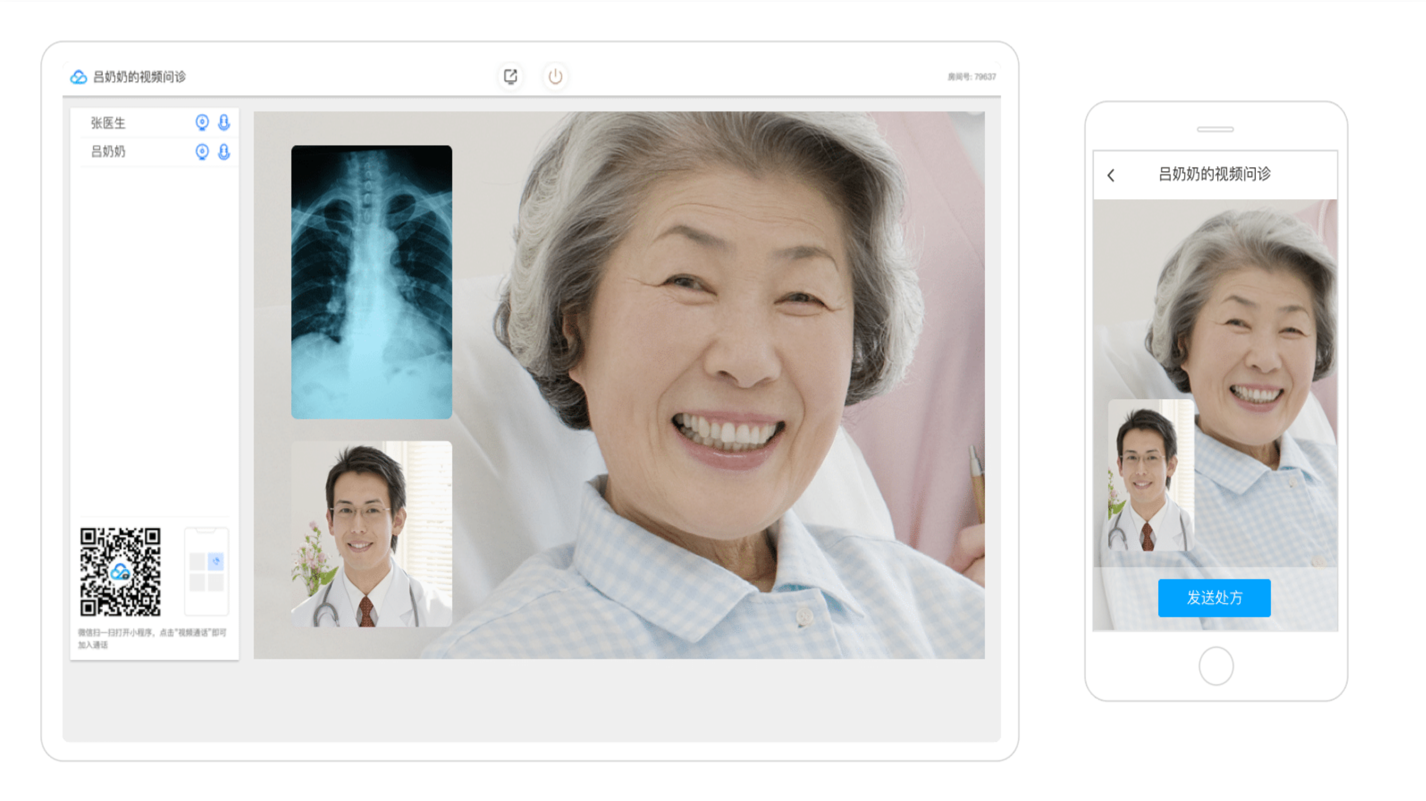
Support for screen sharing, file sharing, and interactive whiteboard features for more efficient online meetings. Combined with instant messaging, it supports various forms of discussion such as text and images without interrupting the meeting process. TRTC offers a network meeting scenario component for direct reuse, significantly reducing development costs. For component usage guidelines, please refer to Video Conferencing.

Search for the required CAM policy as needed, and click to complete policy association.

Online healthcare

Support for ultra-high-definition video resolutions including 1080p, 2K, and 2K+ (specific devices), with flexible video equipment focus adjustment to closely replicate the offline consultation experience. It supports file sharing, screen sharing, and Chat features for sharing medical records and medical imaging, greatly improving consultation efficiency. Support for multi-person video calls allows multiple doctors and patients to participate in online consultations, facilitating smooth medical communication and collaboration.

Search for the required CAM policy as needed, and click to complete policy association.



Video customer service

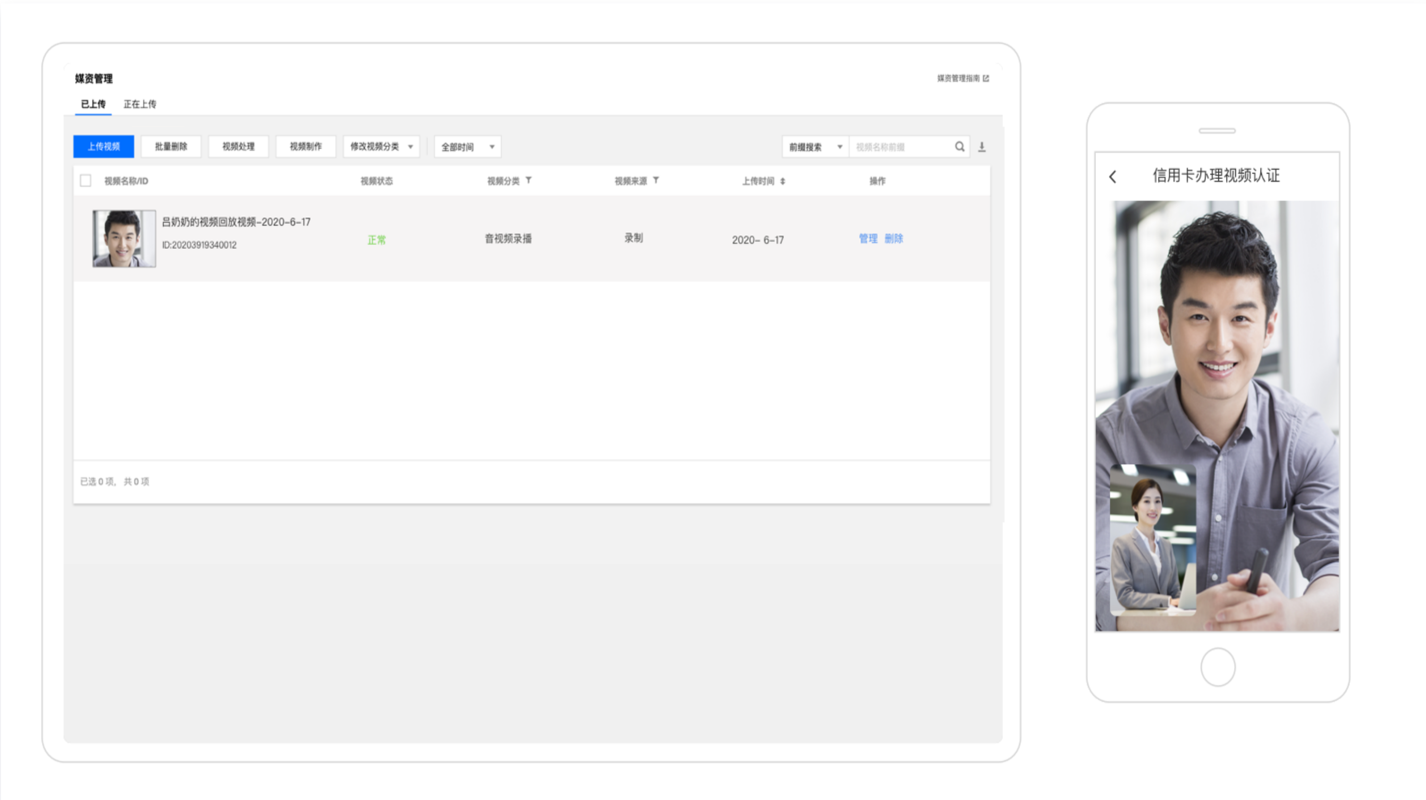
Call latency below 300ms, with packet loss resistance over 70% and network jitter resistance over 1000ms, ensuring smooth and stable calls even in weak network environments. It supports video customer service across mobile apps, PC, mini programs, and web platforms, accessible anytime and anywhere. Support for recording and playback of video service processes greatly improves service quality.

Search for the required CAM policy as needed, and click to complete policy association.

Financial Recording

Provide real-time cloud recording feature covering the entire process, including CFS recording, playback, and download. It also supports server-side deployment for recording to ensure business compliance. Leveraging Tencent's extensive experience in data security over the years, it maximizes data security.

Search for the required CAM policy as needed, and click to complete policy association.



Performance Data

Last updated: 2025-03-25 10:41:41

This document mainly focuses on the key metrics that developers care about most, such as audio and video quality, delay, smoothness, stability, as well as CPU, memory, power consumption, and heating. Objective tests, analyses, and summaries are conducted under **normal, poor network conditions**, and **different real-time interactive scenarios (1v1, 1vN, etc.)**.

Quality Effect in Lossless and Weak Network Environments

Testing Scenario

Audio and video scenarios and voice call scenarios for video calls and online live streaming.

Parameter Configuration

- **Video call:**

Parameter Type	Configuration Message
Resolution	368 × 640
Bitrate	400 Kbps
Frame Rate	15

- **Interactive live streaming:**

Parameter Type	Configuration Message
Resolution	720 × 1280
Bitrate	1200 Kbps
Frame Rate	15

Extreme Network Resistance Testing Data

Extreme network resistance testing refers to testing the maximum network degradation that the SDK can bear in various network degradation environments.

音视频场景的视频抗性极限* (iPhone XR to 小米9)			
	场景	在线直播	视频通话
上行	可承受的最大丢包率	55% loss	65% loss
	可承受的最大网络抖动 (jitter)	1200ms	1700ms
	可流畅通话的最低带宽要求	500kbps	250kbps
	混合损伤条件 (loss + jitter)	20% loss + 300ms	40% loss + 700ms
	混合损伤条件 (loss + delay)	20% loss + 350ms	40% loss + 700ms
下行	可承受的最大丢包率	70% loss	65% loss
	可承受的最大网络抖动 (jitter)	1700ms	1600ms
	混合损伤 (loss + jitter)	40% loss + 800ms	40% loss + 700ms
	混合损伤 (loss + delay)	40% loss + 650ms	40% loss + 600ms

Video resilience limit in audio and video scenarios (iPhone XR to Xiaomi 9)			
Scene		Online live streaming	Video Call
Uplink	Maximum tolerable packet loss rate	55% loss	65% loss
	Maximum tolerable network jitter	1200ms	1700ms
	Minimum bandwidth requirement for smooth calls	500kbps	250kbps
	Composite damage conditions (loss + jitter)	20% loss + 300ms	40% loss + 700ms
	Composite damage conditions (loss + delay)	20% loss + 350ms	40% loss + 700ms

Downlink	Maximum tolerable packet loss rate	70% loss	65% loss
	Maximum tolerable network jitter	1700ms	1600ms
	Composite damage (loss + jitter)	40% loss + 800ms	40% loss + 700ms
	Composite damage (loss + delay)	40% loss + 650ms	40% loss + 600ms

Note:
For specific loss metrics and their meanings, please refer to Appendix 1: Audio and Video Quality Metrics Description.

Audio MOS Value in Poor Network

Data interpretation: Tencent Real-Time Communication (TRTC) can guarantee relatively high sound quality with lower delay in super unsatisfactory network environments.
Following are the objective MOS evaluation results of tencent real-time communication (TRTC) in the following weak network conditions:

音频弱网 MOS 分测试					
场景		Android to Android (小米8 to 小米9)		iOS to iOS (iPhone6 to iPhone6s)	
		score	延迟(ms)	score	延迟(ms)
上行/下行	无损	4.75	186	4.74	209.48
上行	75% loss	3.82	570.7	3.82	554
	2000 jitter	4.28	1362	4.32	1460
	55% loss + 1200 jitter	3.59	1570	3.57	1599
下行	70% loss	4.03	552	4.08	640.9
	2000 jitter	3.53	1584	3.68	1589.78
	50% loss + 900 jitter	4.04	1392	3.93	1418

Audio MOS Test in Poor Network					
Scene		Android to Android (Xiaomi 8 to Xiaomi 9)		iOS to iOS (iPhone6 to iPhone6s)	
		score	Delay (ms)	score	Delay (ms)
Uplink/downlink	Lossless	4.75	186	4.74	209.48
Uplink	75% loss	3.82	570.7	3.82	554
	2000 jitter	4.28	1362	4.32	1460
	55% loss + 1200 jitter	3.59	1570	3.57	1599
Downlink	70% loss	4.03	552	4.08	640.9
	2000 jitter	3.53	1584	3.68	1589.78
	50% loss + 1200 jitter	4.04	1392	3.93	1418

Client SDK Performance Data

Testing Device Info

Device Type	Processor Type	Memory
Android device 1	Snapdragon 625 - 8 cores	4G
Android device 2	Snapdragon 835 - 8 cores	6G
iOS device 1	A8 - dual cores	1G
iOS device 2	A13-6-core	4G

Test Parameter Configuration

Parameter Type	Configuration Message
Resolution	360 × 640
Bitrate	600 kbps

Frame Rate	15

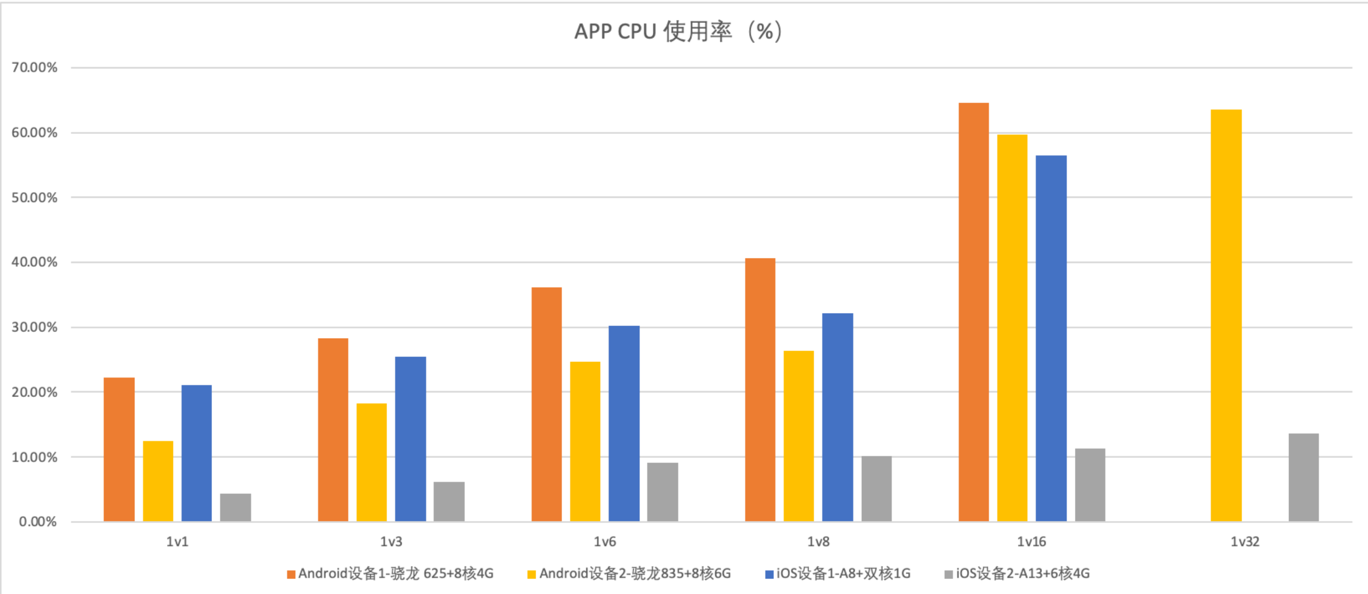
Testing Plan Description

- **Testing Scenarios:** 1v1, 1v3, 1v6, 1v8, 1v16, 1v32.
- **Test duration:** 30 minutes for each scenario.
- **Testing Plan:** Construct multi-person scenarios using Linux streaming robots, with each testing device being independently tested.

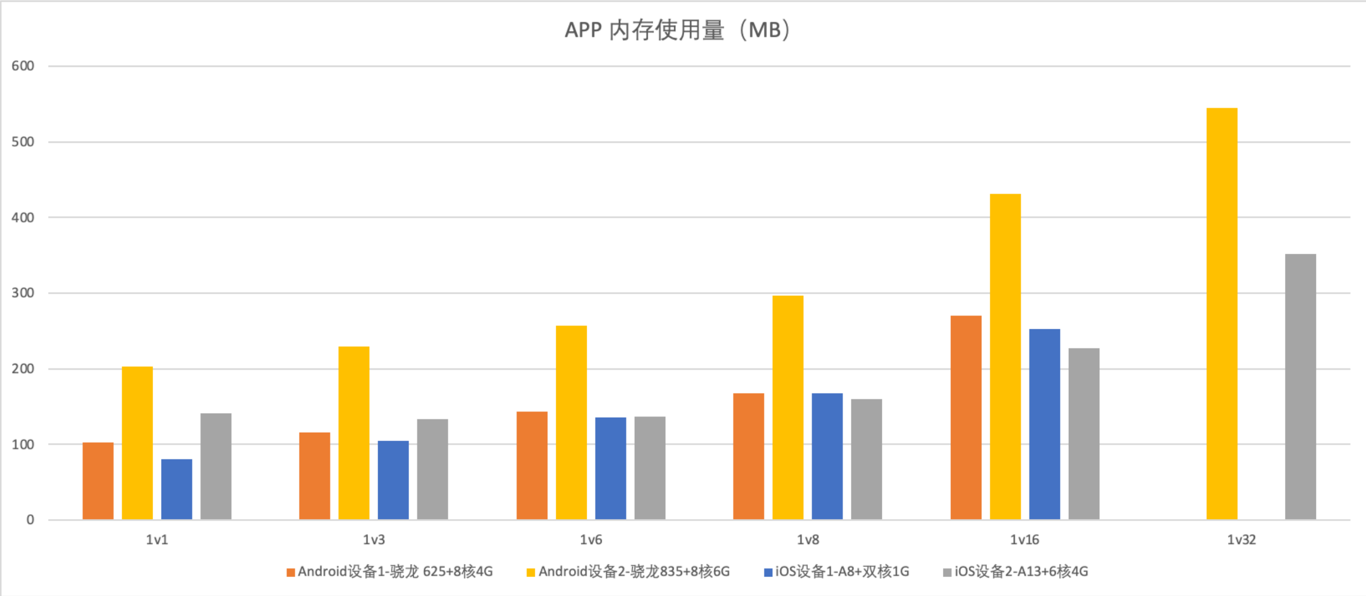
Test Result

Data interpretation: The tencent real-time communication TRTC SDK performs well in various aspects of performance, such as CPU utilization, memory usage rate, heating, and power consumption. It occupies less hardware resources and can provide high-quality audio and video services.
Note: Among them, 1v32 cannot run smoothly on low-end devices, so there is no test data.

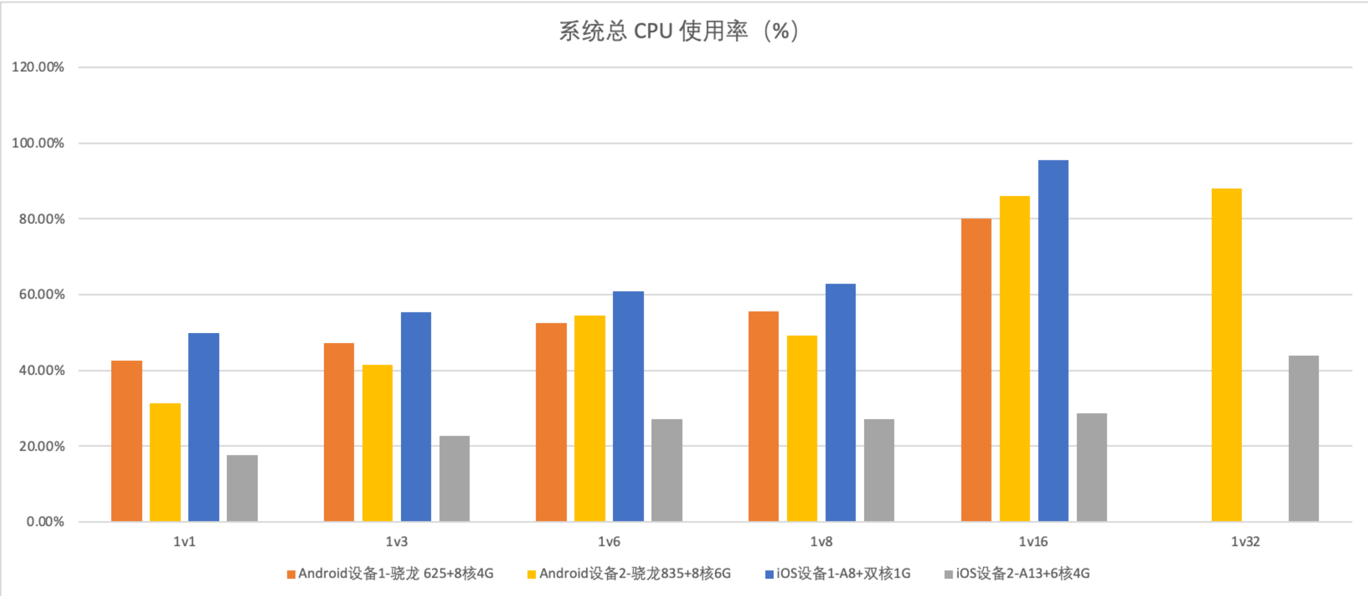
- **App CPU utilization:**



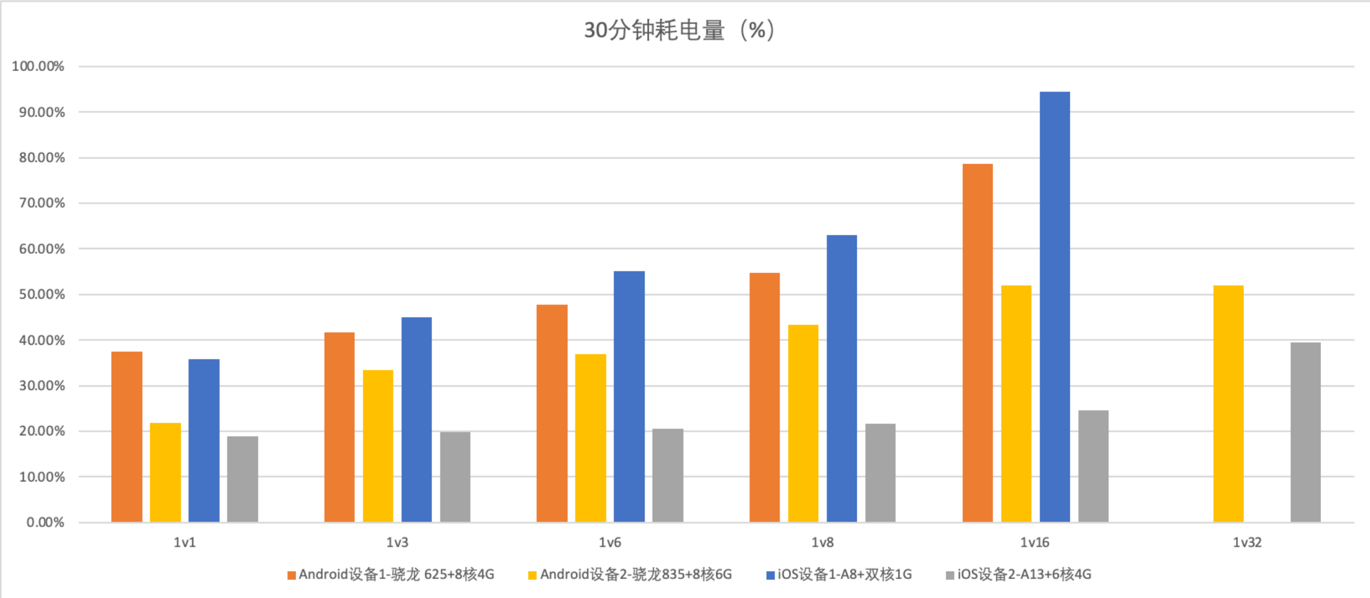
- **App memory usage:**



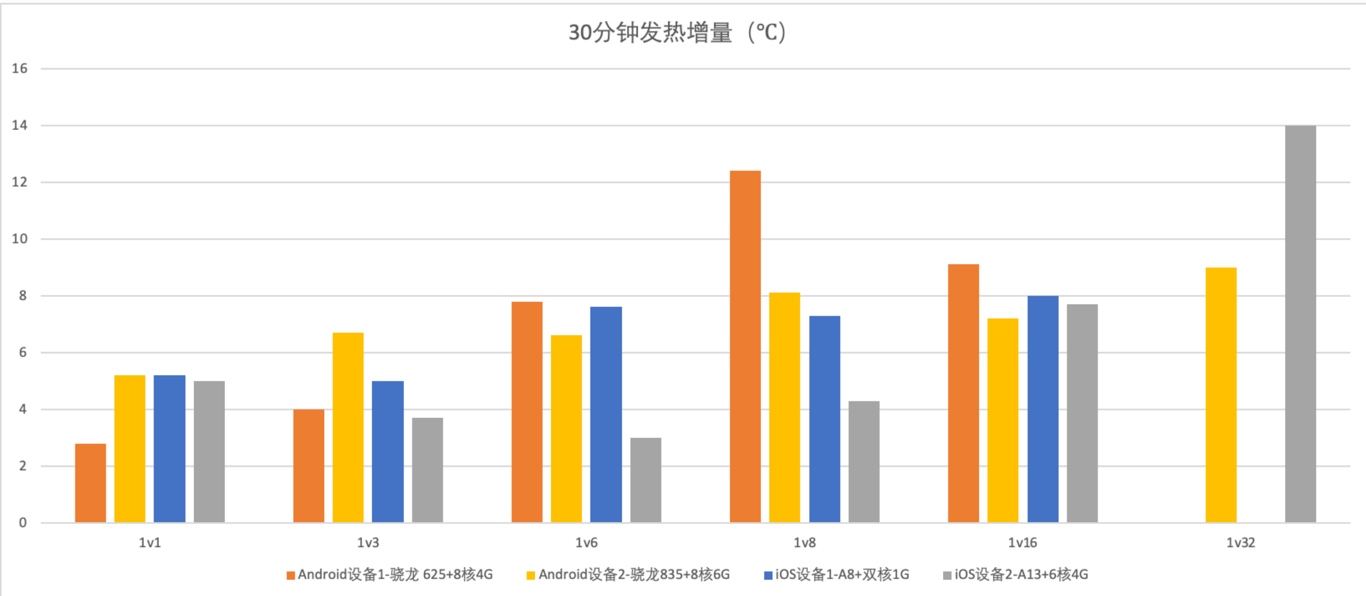
- **Total system CPU utilization:**



- **Battery drain in 30 minutes of running:**



- **Heat increase after running for 30 minutes:**



Appendix 1: Audio and Video Network Impairment Indicator Descriptions

Network Impairment Indicators	Description	Example
Loss	Network	50% Loss means 5 out of 10 packets will be lost

	packet loss	
Delay	Represent latency	200ms Delay means the packets sent by the SDK will be sent over the network after 200ms
Jitter	Represent jitter	300ms Jitter means the packets sent by the SDK may randomly be delayed by 20ms, 280ms, 50ms, 250ms, etc. The maximum delay is 300ms and the average delay is 150ms.

Appendix 2: Data Description of Effect under Poor Network Conditions

Effect Data	Description
MOS value	An important metric commonly used to measure the voice quality of communication systems. The objective MOS value is obtained by using Spirent Nomad to perform POLQA scoring. The higher the score, the better the sound quality.
End-to-end delay	End-to-end latency refers to the time taken from audio capture at the sender to playback at the recipient.
Extreme audio and video resistance testing standard	After adding poor network conditions, use Spirent Nomad separately for POLQA scoring and foreman video sequence transmission. After sending, detect frame interval at the recipient. Observe continuously for more than 10 minutes, collect 30 data points. If there are effects perceived subjectively as abnormal 3 or more times within 3 minutes or an unavailable situation for a long time, it is considered exceeding resistance ability.

Note:
POLQA (Perceptual Objective Listening Quality Assessment) standard, which is scoring based on ITU P.863 International standard, suitable for voice evaluation. POLQA is a globally applicable voice quality analysis standard targeting various network scenes.

Appendix 3: SDK Performance Metric Explanation

Metric Type	Description

App CPU utilization	Android	App CPU indicates the unnormalized CPU utilization of a process. The statistical result is consistent with that of Android Studio Profiler.
	iOS	App CPU indicates the CPU utilization of a process. The statistical result is consistent with that of Xcode. PerfDog usage rate = Xcode usage rate / number of cores.
System CPU utilization	Android	Total CPU indicates the unnormalized CPU utilization of the entire machine. The statistical result is consistent with that of Android Studio Profiler.
	iOS	Total CPU indicates the CPU utilization of the entire machine. The statistical result is consistent with that of Xcode. PerfDog usage rate = Xcode usage rate / number of cores.
Memory Usage	Android	PSS Memory. The statistical result is consistent with the standard result of Android Java API and also consistent with Meminfo.
	iOS	Xcode Memory, statistical method of XCode Debug gauges.
Battery drain	Monitor the battery level during testing. Start recording when the battery level drops from 100% to 99%. Set the ending battery level value and calculate the battery drain in 30 minutes based on the proportion.	
Heat increase	Use a temperature gun to measure the current temperature if the App is not started. Start the App and run it for 30 minutes in each scenario. Heat increase = Temperature after 30 minutes − Temperature when the App is not started.	