Tencent Real-Time Communication

Product Introduction

Product Documentation

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Product Introduction
Overview

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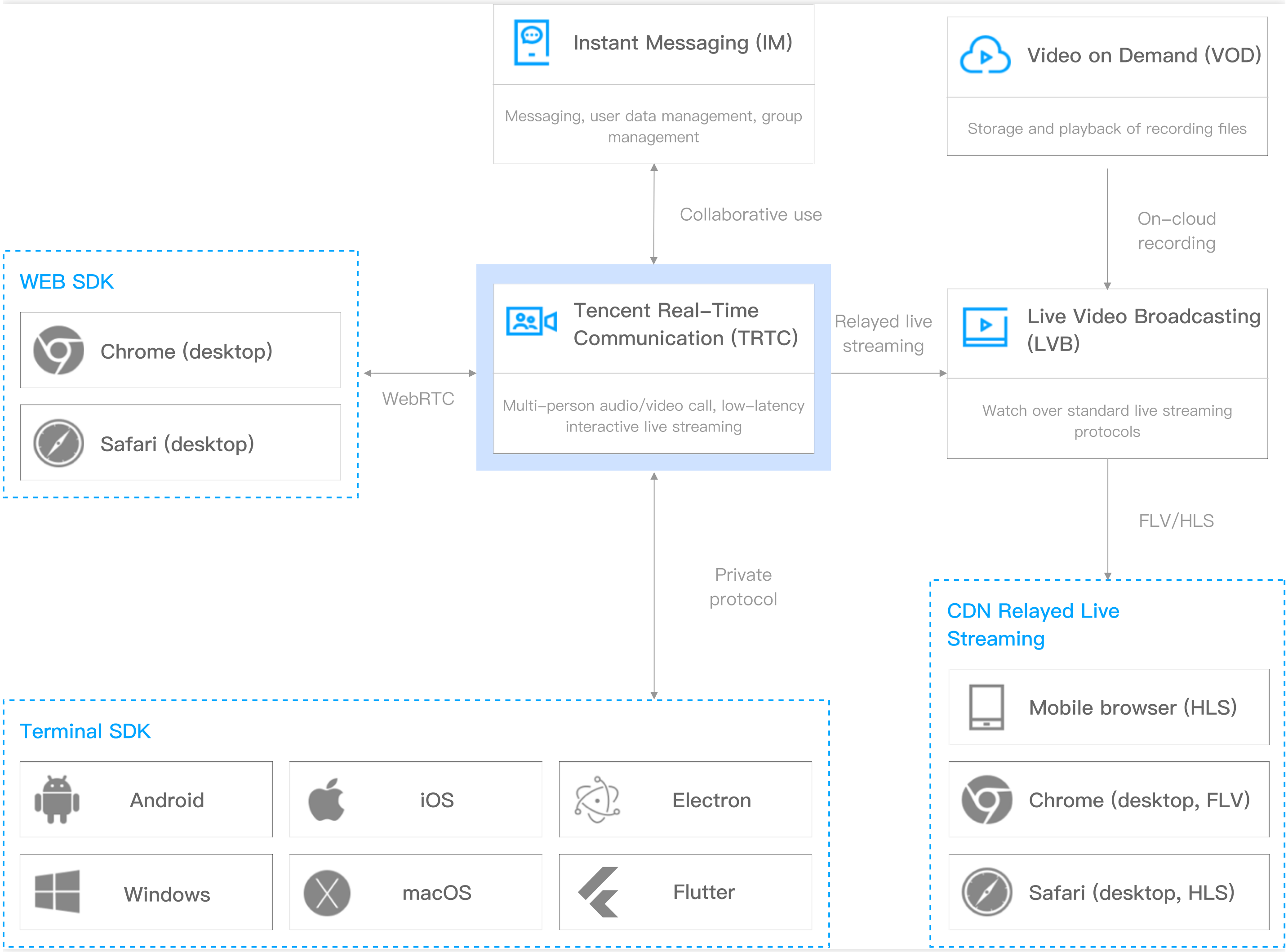
Tencent Real-Time Communication (TRTC) leverages Tencent's 21 years of experience in network and audio/video technologies to offer group audio/video calls and low-latency interactive live streaming solutions, allowing you to quickly develop cost-effective, low-latency, and high-quality interactive audio/video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

- Group audio/video call solution
  Built on top of Tencent Cloud’s Direct Connect network, the solution allows global connection and offers client SDKs and cloud-based APIs for both mobile and desktop platforms. Users can easily access TRTC services via WeChat Mini Programs and webpages too.
- Low-latency interactive live streaming solution
  The 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 lags and a latency below 1 second, taking live streaming into the era of CDN 2.0.

TRTC and Other Tencent Cloud Products

TRTC offers cross-platform solutions for audio/video calls and low-latency interactive live streaming. Through the SDKs TRTC provides for 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 use TRTC in combination with Tencent Cloud’s other products, such as Instant Messaging (IM), Live Video Broadcasting (LVB), and Video on Demand (VOD) to explore more use cases. See the figure below for how TRTC can be used 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	Development Environment
iOS	• iPhone and iPad with iOS 9.0 or above • Xcode 9.0 or above • The project has a valid developer signature.
Android	• Android Studio 3.5 or above • Android 4.1 (SDK API Level 16) or above is recommended.

Platform	Development Environment
Windows	• Windows 7 or above • Visual Studio 2010 or above (Visual Studio 2015 is recommended.) • .Net Framework 4.0 or above
macOS	• Xcode 9.0 or above • Mac computers with OS X 10.10 or above • The project has a valid developer signature.
Web	Desktop Chrome 56 or above is recommended. For more on the development environment, see SDK Quick Integration (Web)
Electron	• Windows 7 or above; OS X 10.10 or above • Electron 4.0.0 or above (the latest Electron SDK is recommended.)
Flutter	iOS: • iPhone or iPad with iOS 9.0 or above • Xcode 9.0 or above • The project has a valid developer signature. Android: • Android Studio 3.5 or above • Android 4.1 (SDK API Level 16) or above is recommended.

Concepts

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

Application

TRTC manages businesses or projects as **applications**. You can create different applications for your businesses or 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 an application is created in the TRTC console. Applications with different `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 user names as `UserID` .
- `UserID` should preferably be 32 bytes or shorter. It can contain digits, letters (case sensitive), and underscores, but must not include only digits.

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 first in order to enter another.

Note :

- The first user who enters a room is the owner of the room. Room owners cannot close rooms manually.
- In the call modes, TRTC closes a room when all users in the room exit.
- In the live streaming modes, if the last user who exits a room is an anchor, TRTC will close the room immediately; if the user is audience, TRTC will close the room in 10 minutes.

- A user will be removed from a room 90 seconds after unexpected disconnection. If all users are unexpectedly disconnected, the room will be closed after 90 seconds.
- If a user attempts to enter a room that does not exist, TRTC will create a room automatically.

Room ID

`RoomID` (room number/ID) uniquely identifies a room in a TRTC application. It is a number in the uint32 range and is assigned and maintained by yourself. Its value range is 1-4294967295.

UserSig

`UserSig` (user signature) is a security signature designed by Tencent Cloud to authenticate user logins, check whether a user is real, and thus prevent attackers from accessing your Tencent Cloud account. For more information, please see FAQs > UserSig.

Push

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

Subscription

Subscription is the operation where 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 and subscribe to and play back the audio/video data of other anchors.
- Audience **can only** subscribe to and play back the audio/video data of anchors.

In the call modes, all users in a room are in the anchor role. In the live streaming modes, users in a room may be divided into anchors and audience. A user can switch roles whenever 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, which are then pushed to a standard live streaming system and distributed through CDNs to audience. For details, see CDN Relayed Live Streaming.

On-cloud recording

TRTC leverages the capabilities of CSS to record entire calls (video/audio) in the cloud and saves the recording files securely in real time in VOD. For details, see On-Cloud Recording and Playback.

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, which 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. For more information, please see On-Cloud MixTranscoding.

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 about the user’s entry or exit.

Features

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Basic Features

Feature	Description	Common Use Cases	Billing
Video call	• One-to-one or group video calls, which support 720p and 1080p definitions • Each room can accommodate up to 300 concurrent users, and up to 50 of them can enable their cameras at the same time.	One-to-one video calls, video conferences with up to 300 attendees, online medical consultation, video chat, video customer service, video interviews, audiovisual recording, online insurance claim settlement, and video Werewolf	Billing of video calls
Audio call	• One-to-one or group audio calls, which support the 48 kHz sample rate and dual channels. • Each room can accommodate up to 300 concurrent users, and up to 50 of them can enable their mics at the same time.	One-to-one or group audio calls, voice chat, audio conferences, audio customer service, and audio Werewolf	Billing of audio calls
Interactive video live streaming	• Cross-room anchor competition • Smooth mic connection/disconnection with no waiting periods; anchor latency below 300 ms • Live streaming to up to 100,000 concurrent viewers and playback latency as low as 1,000 ms in the low-latency live streaming mode • No upper limit on the number of viewers in the CDN relayed live streaming mode	Low-latency video live streaming, interactive classrooms with up to 100,000 participants, live video competitions, video dating, remote training, large-scale conferences, etc.	Billing of interactive video live streaming
Interactive audio live streaming	• Cross-room anchor competition • Smooth mic connection/disconnection with no	Low-latency audio live streaming, live audio co-anchoring, live audio	Billing of interactive

	waiting periods; anchor latency below 300 ms • Live streaming to up to 100,000 concurrent viewers and playback latency as low as 1,000 ms in the low-latency live streaming mode • No upper limit on the number of viewers in the CDN relayed live streaming mode	competitions, voice chat rooms, audio dating, karaoke rooms, FM radio, etc.	audio live streaming

Advanced Features

Feature	Description	Common Use Cases	Billing
Cross-room competition	Anchors from different rooms compete with each other while viewers watch.	Live show streaming, cross-room teaching, etc.	You will be charged basic service fees for using this service.
Screen sharing	Sharing the desktop, a window (e.g., a PowerPoint playback window), or a desktop section of the local user to others	Online classrooms, slide sharing, remote assistance, etc.	You will be charged basic service fees for using this service.
On-cloud recording	TRTC leverages the capabilities of CSS and the relayed push technology to offer on-cloud (audio/video) recording throughout a call. Recording files are saved reliably and in real time in VOD.	Audiovisual recording, archiving, compliance, etc.	On-cloud recording is a value-added service, for which you will be charged an additional on-cloud recording fee
Local server recording	(Audio/video) recording on a local server. To try this feature out, contact us for the SDK and instructions.	Audiovisual recording, archiving, compliance, etc.	You will be charged basic service fees for using this feature.

Feature	Description	Common Use Cases	Billing
High audio quality	• High audio quality at 48 kHz sample rate • Stereo with left and right audio channels, comparable in audio quality to CDs	Audio calls, video calls, interactive live streaming, voice chat rooms, high-quality FM radio, music classes, karaoke rooms, online classrooms, etc.	Free
High image quality	720p and 1080p HD videos	Video calls, interactive live streaming, online classrooms, etc.	Free
3A processing	TRTC uses the industry-leading Tencent Real-time Audio Engine (TRAE) for acoustic echo cancellation (AEC), active noise suppression (ANS), and automatic gain control (AGC) to deliver better audio quality when multiple people speak at the same time or in the presence of background noises.	All audio scenarios	Free
Basic beauty filters	Basic beauty filters such as skin brightening, skin smoothing, and rosy complexion	Video calls, interactive live streaming, online classrooms, etc.	Free
Background music	Using local music files in the formats of MP3, AAC, WAV, and others as background music	Audio calls, video calls, interactive live streaming, online classrooms, voice chat rooms, karaoke rooms, FM radio, etc.	Free
Audio effects	Adding audio effects such as applauding, cheering, whistling, and booing during a call	Audio calls, video calls, interactive live streaming, voice chat rooms, karaoke rooms, FM radio, etc.	Free

Feature	Description	Common Use Cases	Billing
Local background audio	Sending audio played locally, for example, the music played via QQ Music on the local user’s computer, to others	Interactive live streaming, online classrooms, voice chat rooms, FM radio, etc.	Free
Voice changing	Voice effects such as little girl, middle-aged man, and heavy metal	Audio calls, video calls, interactive live streaming, voice chat rooms, karaoke rooms, FM radio, etc.	Free
Reverb	Reverb effects such as karaoke room, small room, concert hall, and bathroom	Audio calls, video calls, interactive live streaming, voice chat rooms, karaoke rooms, FM radio, etc.	Free
Volume callback	Showing volume in waveform animations or via prompts	Audio calls, video calls, voice chat rooms, FM radio, karaoke rooms, voice activity detection, etc.	Free
In-ear monitoring	Recording local audio and playing it back in the local user’s audio, usually for detection of speech errors or pitch control during singing	Interactive live streaming, live show streaming, karaoke rooms, etc.	Free
Custom audio data	Calling back raw audio for custom processing. You can connect the SDK to non-standard external devices or use audio files, etc.	Non-standard device connection, custom audio effect, speech processing, speech recognition, etc.	Free

Feature	Description	Common Use Cases	Billing
Custom video data	Custom video sources and renderers. Non-camera video sources such as video files, external devices, and third-party custom data sources can be used.	Custom beauty filters, custom data sources, multi-device management, video recognition, image processing, etc.	Free
SEI information	Embedding custom information such as lyrics and questions as SEI frames into video streams	Karaoke rooms, live quizzes, interactive live streaming, etc.	Free

Extended Features

Note :

Extended features are value-added services provided by TRTC in collaboration with other Tencent Cloud products and are charged according to the billing standards of the corresponding products.

Feature	Description	Common Use Cases	Billing
CDN live streaming	TRTC uses relaying and transcoding clusters to convert its UDP audio/video streams into RTMP streams in the cloud, which are then pushed to the standard live streaming system and distributed through CDNs to viewers.	Interactive live streaming, live sharing, large-scale conferencing, live stream watching by remote viewers, etc.	Relayed live streaming is a value-added service and is charged by **CSS**. For more information, please see CDN Relayed Live Streaming > Applicable Fees.

Feature	Description	Common Use Cases	Billing
Instant messaging (IM)	• TRTC leverages the capabilities of IM, including one-to-one chat, group chat, and chat rooms with no upper limit on user number, to enable features such as chatting, commenting, and sending on-screen comments, gifts, and likes. • IM can also be used for signaling-based interaction, call making, and user number counting.	Online customer service, interactive live streaming, interactive classrooms, remote training, etc.	IM is a value-added service and is charged by **IM**. For more information, please see IM > Purchase Guide > Pricing.
Speech content moderation	Detecting pornographic, politically sensitive content, etc. for content-related risk management	Business security protection, compliance, etc.	Speech content moderation is a value-added service and is charged by **Business Security Protection (BSP)**. To try it out, contact us to activate the service.
Video content moderation	Detecting pornographic, politically sensitive content, etc. for content-related risk management	Business security protection, compliance, etc.	Video content moderation is a value-added service and is charged by CSS. For more information, please see Intelligent Porn Detection.

Strengths

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Cross-platform Global Interconnection

Implementing authentic cross-platform compatibility in the industry, TRTC is perfectly compatible with over 5,000 device models. It provides client SDKs and TencentCloud APIs for iOS and Android mobile devices and Windows/macOS desktops to make it easy for users to access the TRTC service for global interconnection on WeChat, QQ, WeChat Work, WeChat Mini Program, and desktop browsers.

Low-Threshold Quick Integration

You can run the TRTC demo with merely two lines of code, integrate the general capabilities of TRTC with ten lines of code, and build a low-latency, low-lag, and high-quality real-time audio/video interactive product from scratch within 1 minute at the soonest. For detailed directions, please see Quick Demo Run and Quick SDK Integration.

Scenario-based Custom Components

TRTC provides a rich set of scenario-based custom components to help you quickly implement various features in the simplest way, such as voice chat, conferencing, interactive live streaming, and interactive teaching. For detailed directions, please see Use Cases.

Low Latency

TRTC offers a highly connected, reliable, and secure network across the globe. Leveraging our proprietary multi-addressing algorithms, TRTC has the ability to stream users' audio/video data to optimal nodes across the entire network. With abundant high-bandwidth resources and globally-distributed edge servers, it can ensure **an average end-to-end latency of below 300 ms** between countries/regions.

Low Lag

TRTC reduces lags through intelligent QoS and optimized encoding, and is resilient against **a packet loss rate of over 80%** and **network jitter of over 1,000 ms**. Even with poor network condition, it can ensure a high-quality and stable audio/video communication.

High Quality

TRTC supports 720p and 1080p HD video and enables you to have smooth video calls even at a 70% packet loss rate. **It also supports 128 kHz high-quality audio** and enables you to make clear audio calls even at a 80% packet loss rate. In addition, leveraging its industry-leading 3A processing technologies (i.e., acoustic echo cancellation (AEC), active noise suppression (ANS), and automatic gain control (AGC)), it eliminates echoes and howling, rendering a lossless audio quality comparable to that of CDs.

Use Cases

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Tencent Real-Time Communication (TRTC) features two major solutions: low-latency interactive live streaming and audio/video call. It has a wide variety of capabilities, such as low-latency live streaming, real-time recording, screen sharing, beauty filters, and stereo sound, and can be seamlessly connected to CDN. It is ideal for business scenarios like interactive co-anchoring, cross-room competition, radio, karaoke, small/big online class, voice chat, video chat, and online conferencing. This document describes the business scenarios covered by the two major solutions.

Interactive Audio Live Streaming

Voice chat room

TRTC allows up to 50 users to mic on and chat at the same time and smoothly mic on/off with a latency below 300 ms, and supports multiple audio effects such as voice changing, ambient sound effects, and reverb, enriching and diversifying the voice chat experience. It can be integrated with Tencent Cloud IM to support various message interaction methods such as public chat, private chat, group chat, liking, and gifting, delivering an excellent interactive chat experience. In addition, it provides a scenario-based voice chat room component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Voice Chat Room.

Radio

TRTC supports 48 kHz sample rate, 128 Kbps bitrate, and stereo sound, delivering an authentic sound experience comparable to CD. It can play back local music in various formats such as MP3, AAC, and WAV as the background music, allowing you to easily build a high-quality radio station. It offers a rich set of voice changing effects like middle-aged man and little girl, making the radio more entertaining. In addition, it provides a scenario-based radio component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Voice Chat Room.

Online karaoke

TRTC supports 48 kHz sample rate, 128 Kbps bitrate, and stereo sound, delivering a smooth online karaoke experience comparable to recording studio. It features an ultra-low latency below 300 ms, making it ideal for multi-person singing. It provides multiple sync mechanisms such as message passthrough and timestamp, helping precisely align the accompaniment, vocal, and lyrics during online karaoke. It also supports in-ear monitoring to help users carry a tune with ease.

Interactive Video Live Streaming

Show live streaming

TRTC delivers a latency below 300 ms for cross-room co-anchoring competition and enables viewers to co-anchor with the anchor and mic-on/off smoothly, which help meet the frequent interaction needs in show live streaming scenarios. It supports smart beauty filters, making the show more charming. In addition, it provides a scenario-based show live streaming component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Video Interactive Live Streaming.

Interactive big class

TRTC allows 100,000 students to watch at the same time with a latency below 300 ms and enables teacher-student co-anchoring and smooth mic-on/off for a favorable class interaction experience. It supports various class application features such as screen sharing, interactive whiteboard, and recording and playback, enriching and diversifying teaching methods in interactive online big classes. In addition, it provides a scenario-based interactive big class component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Real-Time Interactive Teaching.

Interactive small class

TRTC supports interactive small classes in multiple sizes such as **1-to-1**, **1-to-2**, **1-to-6**, and **1-to-32**, where the teacher can interact with students with a latency below 300 ms, delivering a smoother teaching experience. It offers various class application features like screen sharing, courseware sharing, and interactive whiteboard, enriching and diversifying the online teaching methods. The entire lecture can be recorded and then played back on demand later, helping enhance the learning results.

Mini Program live streaming

TRTC supports interconnection across Mini Program, application, and PC, so the live streaming content can be quickly delivered to users on different platforms. By integrating with features such as instant messaging, video on-demand, and UGSV, it supports a diversity of features like live room on-screen commenting, liking, gifting, LVB recording, and recording playback, which improve the Mini Program live streaming experience. Moreover, it is capable of intelligently detecting porn in videos and images, which can process non-compliant contents within seconds and ensure content compliance and business security.

Q&A live streaming

TRTC supports low-latency live streaming under high concurrence, which minimizes the viewers' watch latency and is suitable for displaying questions synchronously on viewer clients. It supports multiple sync mechanisms such as message passthrough, timestamp, and signaling channel so as to precisely display the audio, images, and questions in sync, meeting the ultra-high-concurrence IM interaction needs. It also supports quiz interaction, result statistics collection, and co-anchoring in live streaming, making online Q&A more interesting. Plus, it can filter keywords and answer hints during interactive live streaming in real time, which helps improve the user experience and reduce the risks of business non-compliance.

Audio Call

Multi-person audio call

TRTC allows 300 users to call at the same time and up to 50 of them to mic on at the same time. With a sample rate of 48 kHz, a bitrate of 128 Kbps, and integration with Tencent Cloud's outstanding 3A processing technologies, it can deliver a smooth and high-quality audio call experience. In addition, it provides a scenario-based multi-person audio call component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Real-Time Audio Call.

One-to-One audio call

With a call latency of less than 300 ms, packet loss prevention rate of over 80%, and network jitter prevention of over 1,000 ms, TRTC can deliver a smooth and stable one-to-one audio call experience even in weak network environments. By integrating with the rich variety of call signaling management APIs provided by Tencent Cloud IM, it is ideal for various audio call scenarios. In addition, it provides a scenario-based one-to-one audio call component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Real-Time Audio Call.

Werewolf

With a call latency of less than 300 ms, packet loss prevention rate of over 80%, and network jitter prevention of over 1,000 ms, TRTC guarantees smooth and stable werewolf gameplay even in weak network environments. It supports real-time monitoring of user network status in so as to ensure an outstanding gaming experience. Plus, it can test user audio devices to eliminate exceptional muting and further improve the gaming experience.

Audio conferencing

TRTC supports cross-platform compatibility, enabling users to flexibly join meetings on mobile phones, PCs, tablets, and WeChat. By leveraging its outstanding 3A processing technologies, it can eliminate echoes and howling, ensuring smooth and clear conferencing. It also supports interactive whiteboard and file sharing, making communication in audio conferencing more efficient.

Video Call

Multi-person video call

TRTC supports multi-person video call and provides HD image quality of 720p and 1080p, with one single room able to sustain up to 300 concurrent online users and allow 50 users to enable video at the same time. It has diverse features such as instant messaging, video on-demand, recording, and porn detection, making it ideal for a lot of use cases. In addition, it provides a scenario-based multi-person video call component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Real-Time Video Call.

One-to-One video call

TRTC supports one-to-one video call and provides HD image quality of 720p and 1080p, delivering a high-quality video call service. It has diverse features such as instant messaging, screen sharing, recording, and interactive whiteboard, making it ideal for a lot of use cases. In addition, it provides a scenario-based one-to-one video call component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Real-Time Video Call.

Online conferencing

TRTC supports conferencing application features such as screen sharing, file sharing, and interactive whiteboard, making online conferencing more efficient. By integrating with Tencent Cloud IM, it offers multiple discussion methods like text/image-based communications without interrupting the conferencing process. In addition, it provides a scenario-based online conferencing component, which can be directly reused to minimize the development costs. For more information on how to use the component, please see Video Conferencing.

Online healthcare

TRTC supports 1080p FHD image quality and allows you to flexibly adjust the video device focus, which is highly comparable to the medical diagnosis experience in hospitals. It has a lot of convenient features such as file sharing, screen sharing, and instant messaging for medical record and image sharing, greatly improving the diagnosis efficiency. Moreover, it supports multi-person video call, where multiple physicians and patients can join an online diagnosis session, helping deliver a smooth healthcare communication and collaboration experience.

Video customer service

With a call latency of less than 300 ms, packet loss prevention rate of over 70%, and network jitter prevention of over 1,000 ms, TRTC guarantees smooth and stable call communication even in weak network environments. It enables interconnection across various platforms such as mobile application, PC, Mini Program, and web, making the video customer service accessible any time, any where. Plus, it can record and play back the video service process, which helps greatly improve the service quality.

Financial audiovisual recording

TRTC provides a real-time on-cloud recording feature throughout the entire call that supports recording file storage, playback, and download as well as deployment on local servers, helping ensure your business compliance. In addition, by leveraging Tencent's 21 years of experience in data security, it can guarantee data security at its best.