Tencent Real-Time Communication

Product Introduction

Product Documentation

Contents

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 or above. For more on the development environment, see SDK Quick Integration (Web).
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 memebers. 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 CDN Relayed Live Streaming.

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 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. 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. For more information, 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 when the user enters or exits the room.

Features

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Basic Features

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

	their mics on at the same time. 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 No upper limit on audience size in the CDN live streaming mode		
Interactive audio streaming	Audio communication between anchors and audience Cross-room anchor communication Smooth mic on/off without waiting; anchor latency less than 300 ms 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. 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 No upper limit on audience size in the CDN live streaming mode	Low-latency live audio streaming, live audio co-anchoring, live audio competitions, audio chat rooms, audio dating, karaoke rooms, FM radio	

Advanced Features

Feature	Description	Common Use Cases	Billing

Co-anchoring	Smooth mic on/off without waiting	Interactive live streaming, online class, chat room	Basic service fees are charged for using the co-anchoring feature.
Cross-room communication	Anchors from different rooms can communicate with each other while audience members watch.	Showroom streaming, cross-room communication, cross-room class	Basic service fees are charged for using the cross-room communication feature.
Screen sharing	Sharing the desktop, a window (for example, a Microsoft PowerPoint window), or a portion of the desktop	Online class, slideshow sharing, remote assistance	Basic service fees are charged for using the screen sharing feature.
Server-side local 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.	Audiovisual recording, archiving, compliance	Basic service fees are charged for using the server-side local recording feature.
On-cloud recording	On-cloud recording relies on the relay-to-CDN feature and leverage the capabilities of CSS to record live streaming sessions (audio/video). Recording files are saved securely and in real time to VOD.	Audiovisual recording, archiving, compliance	On-cloud recording is a value-added service and incurs additional fees.
On-Cloud MixTranscoding	TRTC 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.	Stream mixing, recording format conversion	On-Cloud MixTranscoding is a value-added service and incurs additional fees.
High audio quality	48 kHz sample rate, end-to-end 192 Kbps bitrate, and dual channels for a clear and	Audio call, video call, interactive live	Free of charge

	immersive audio interaction experience	streaming, audio chat room, high-audio-quality FM radio, music class, karaoke, online class	
High video quality	720p/1080p HD video	Video call, interactive live streaming, online class	Free of charge
3A processing	Leveraging the industry-leading 3A (acoustic echo cancellation, active noise suppression, automatic gain control) technologies of Tencent Ethereal Audio Lab, TRTC can ensure audio quality even when multiple people speak at the same time or in the presence of background noises.	All audio scenarios	Free of charge
AI-based noise suppression	Removing inconstant noises that traditional noise suppression technologies cannot handle, such as cough, sneeze, and car horn	Audio call, video call, interactive live streaming, audio chat room, online class	Apply for beta testing
Basic beauty filters	Basic beautification features, including skin brightening, skin smoothing, blush, and basic filters	Video call, interactive live streaming, online class	Free of charge
Background music	Using local music files in formats such as MP3, AAC, and WAV as background music	Audio call, video call, interactive live streaming, online class, audio chat room, karaoke, FM radio	Free of charge
Audio effects	Audio effects such as applauding, cheering, whistling, and booing	Audio call, video call, interactive live streaming, audio chat room, karaoke, FM radio	Free of charge
Publishing system audio	Publishing the audio you play locally, for example, the music played by QQ Music on your computer, to remote users	Interactive live streaming, online	Free of charge

		class, audio chat room, FM radio	
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 of charge
Reverb	Reverb effects such as karaoke, small room, hall, and bathroom	Audio call, video call, interactive live streaming, audio chat room, karaoke, FM radio	Free of charge
Volume callback	Callback of audio volumes, which you can use to display audio waveforms or volume reminders	Audio call, video call, audio chat room, FM radio, karaoke, and speech detection	Free of charge
In-ear monitoring	Recording local audio and playing 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 of charge
Custom audio data	Callback of raw audio for custom processing. You can connect the SDK to non-standard external devices or use local audio files.	Non-standard device connection, custom audio effect, speech processing, speech recognition	Free of charge
Custom video data	Custom video sources and renderers. You can use non-camera video sources such as video files, external devices, and third-party sources.	Custom beauty filters, custom data sources, multi-device management, video recognition, image processing	Free of charge
SEI message	Embedding custom information such as lyrics and questions as SEI frames into published video streams	Karaoke, live quiz, interactive live streaming	Free of charge

Extended Features

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Extended features are provided by 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 the audience.	Interactive live streaming, live sharing, large-scale conferencing, live stream viewing by remote users	CDN live streaming is charged by CSS.
IM	You can use the capabilities of IM, including one-to-one chat, group chat, and chat rooms with no upper limit on user number, to implement features such as chat, comments, and sending on-screen comments, gifts, and likes. You can also use IM for signaling-based interaction, call invitations, and user counting.	Video customer service, interactive live streaming, interactive classroom, remote training	IM features are charged by IM. For details, see Pricing.
AI-based 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	AI-based beauty filters are charged by Tencent Effect SDK.

Strengths

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Cross-platform global connection

TRTC is a cross-platform solution compatible with more than 5,000 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 IM. 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 IM 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 Auido/Video Call.

Online conferencing

Screen sharing, file sharing, and interactive whiteboard for a more efficient conferencing experience; text/image chat enabled by Tencent Cloud IM. 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

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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:

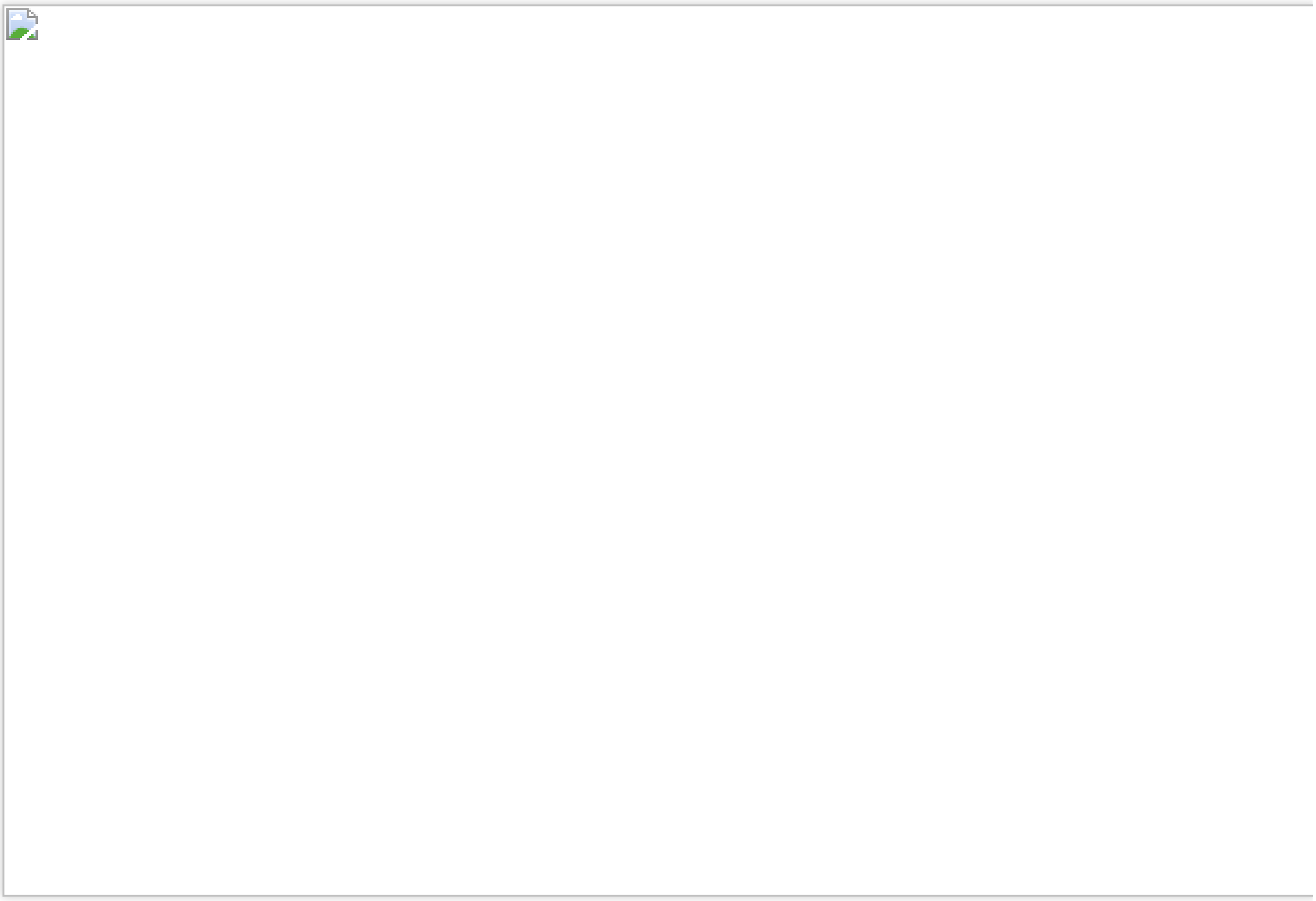
Parameter	Value
Resolution	368 x 640
Bitrate	400 Kbps
Frame rate	15

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.



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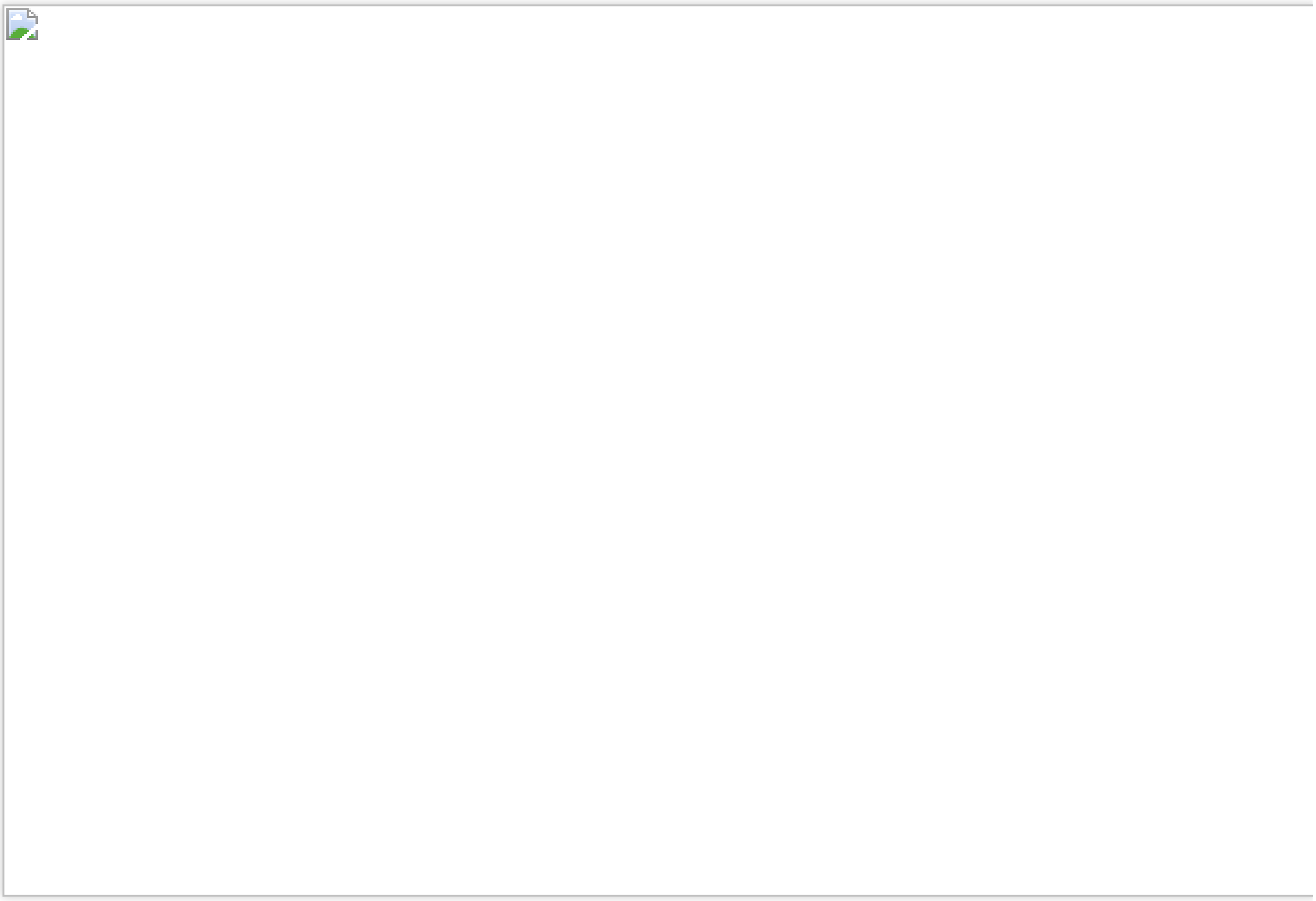
For the metrics used to measure tolerance to poor network conditions, please see Appendix 1: Network Metrics.

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.



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

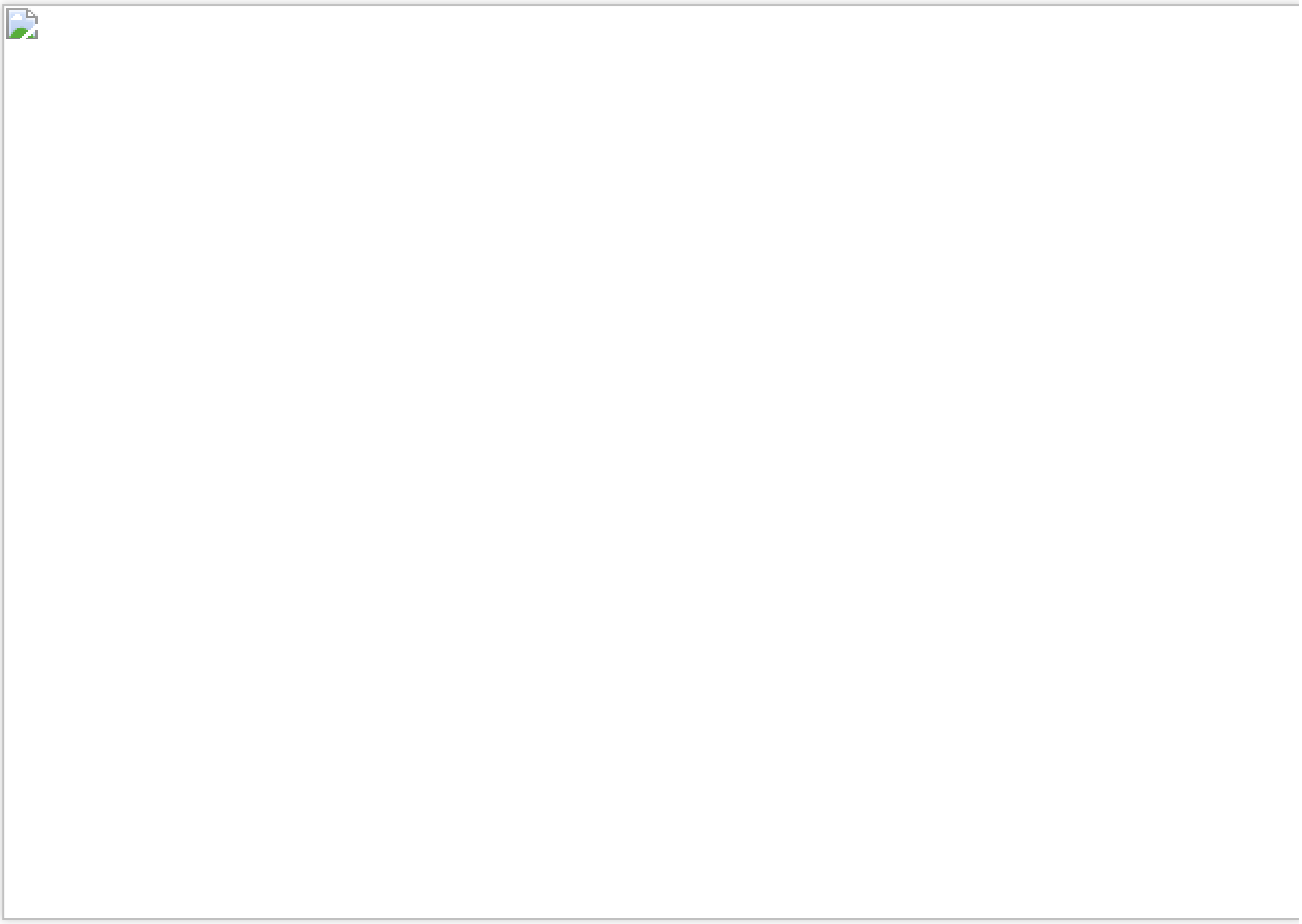
Duration: 30 min for each scenario

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

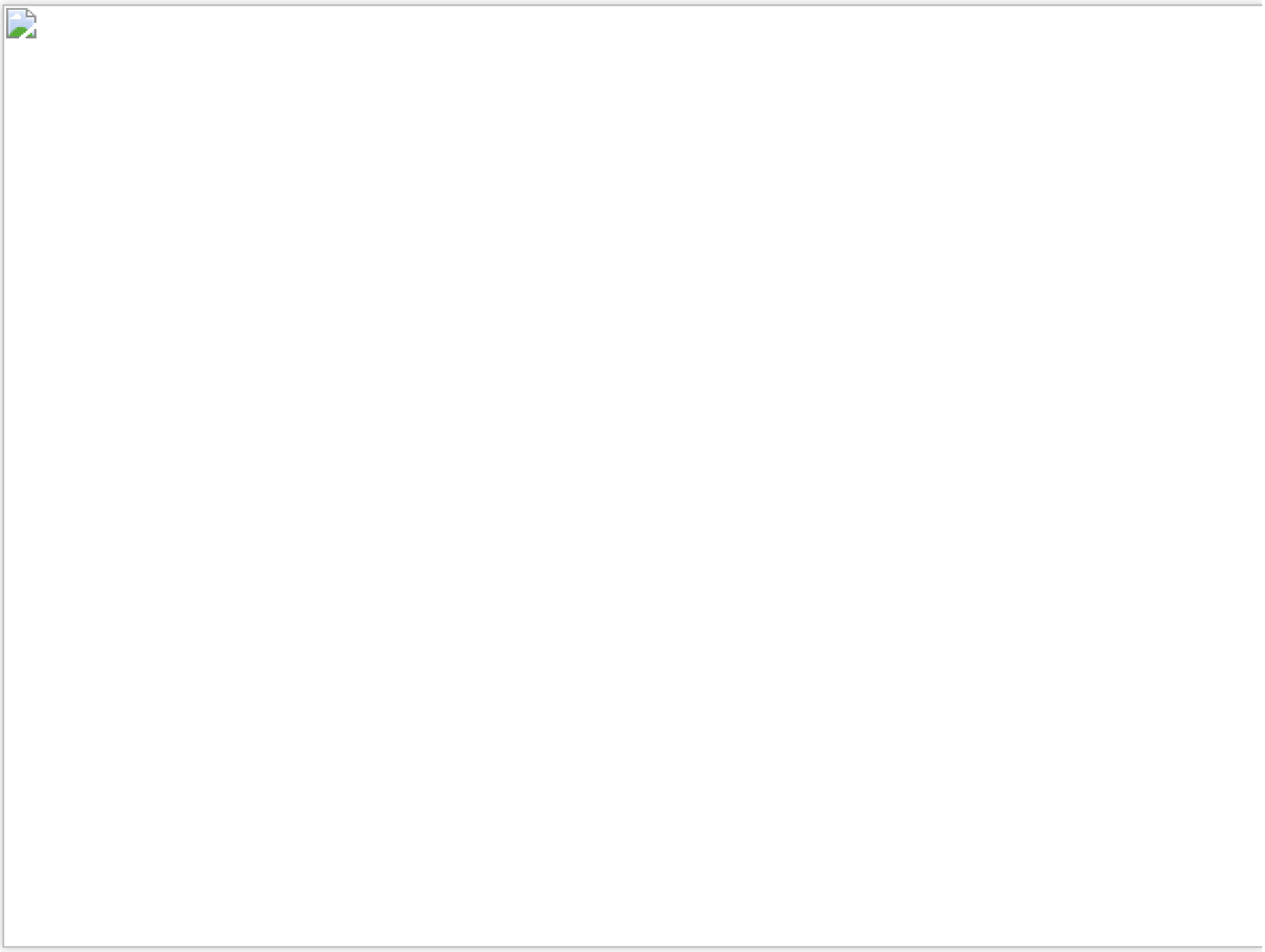
Test result

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.

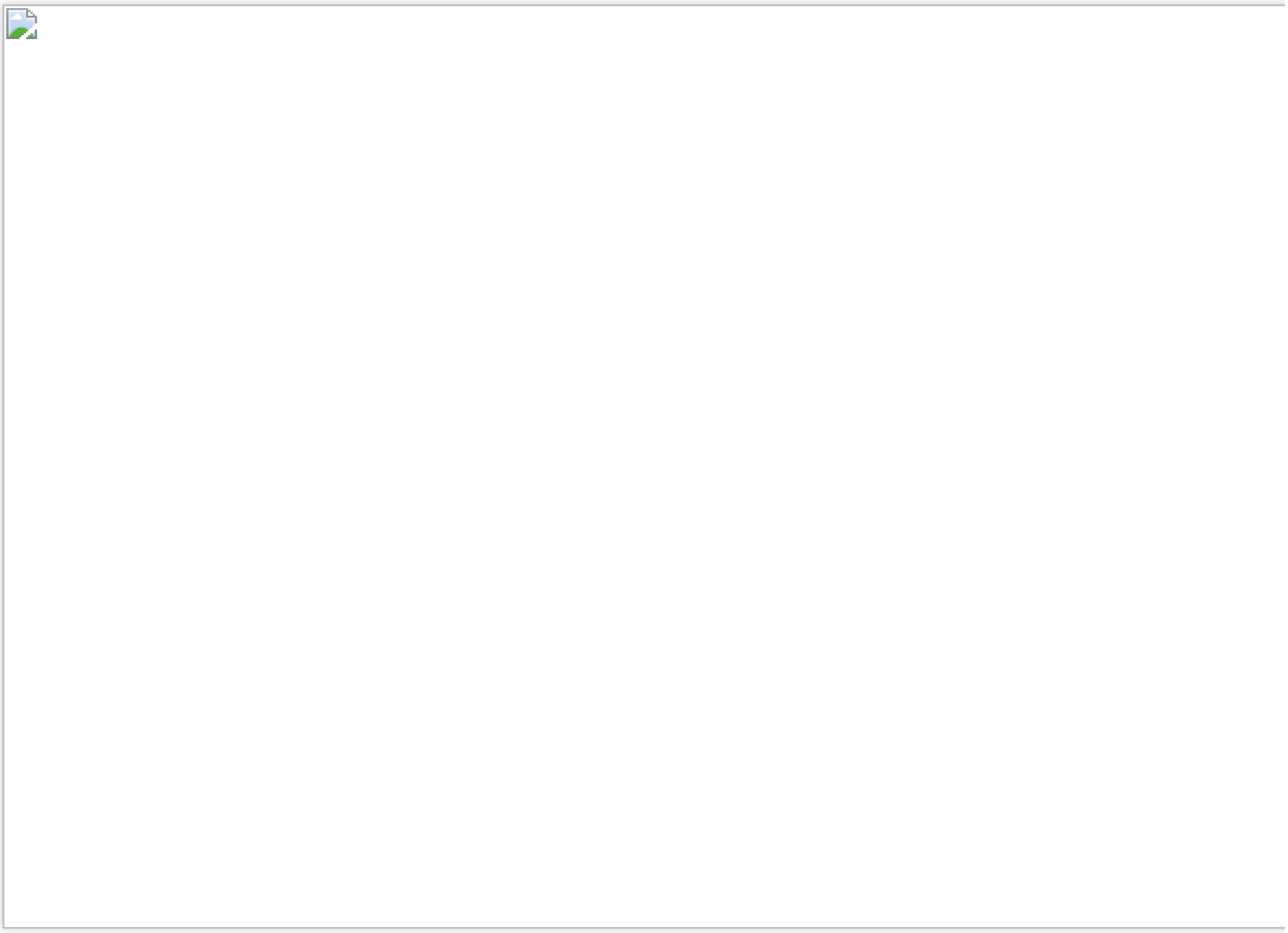
App CPU usage:



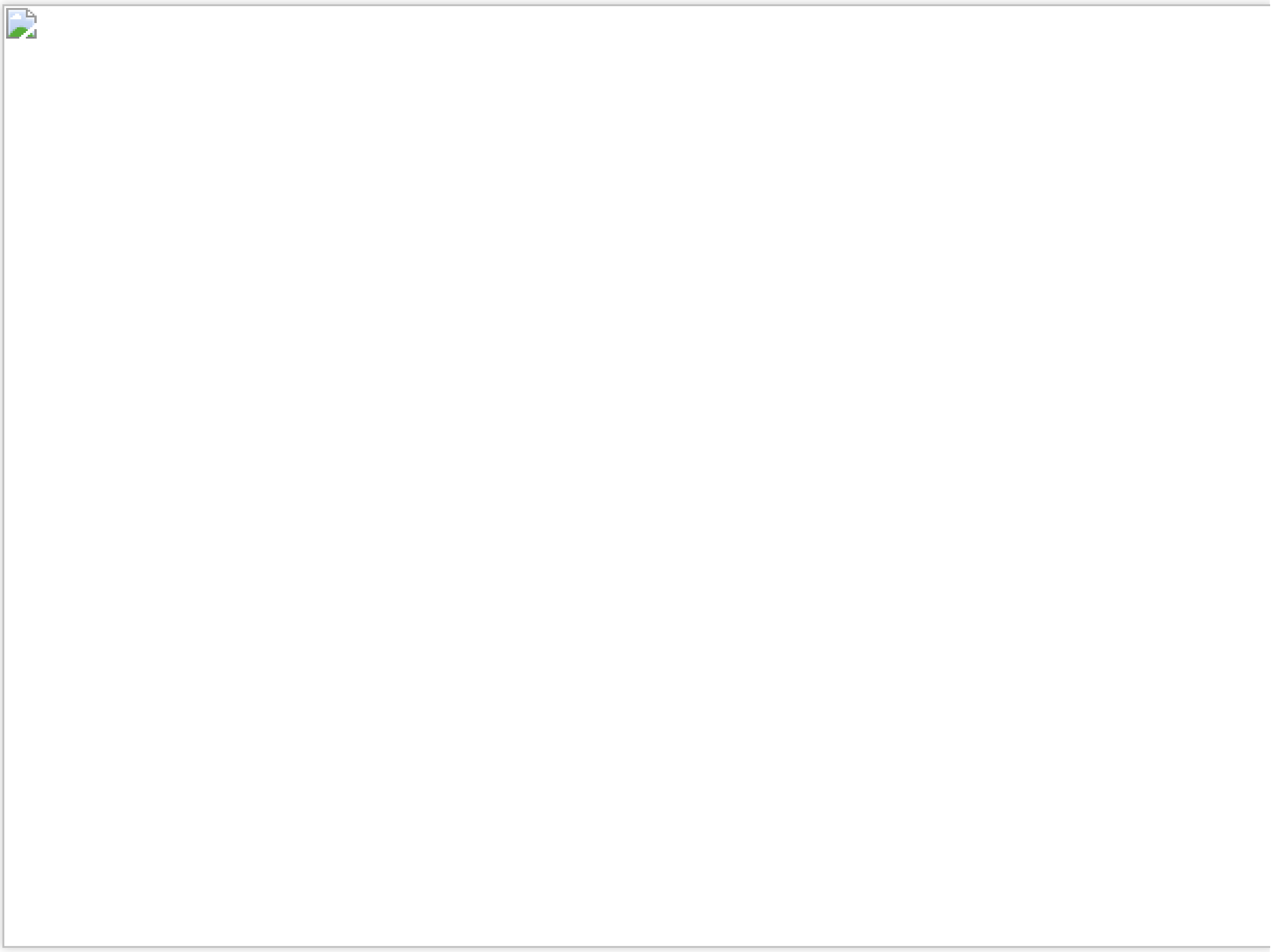
App memory usage:



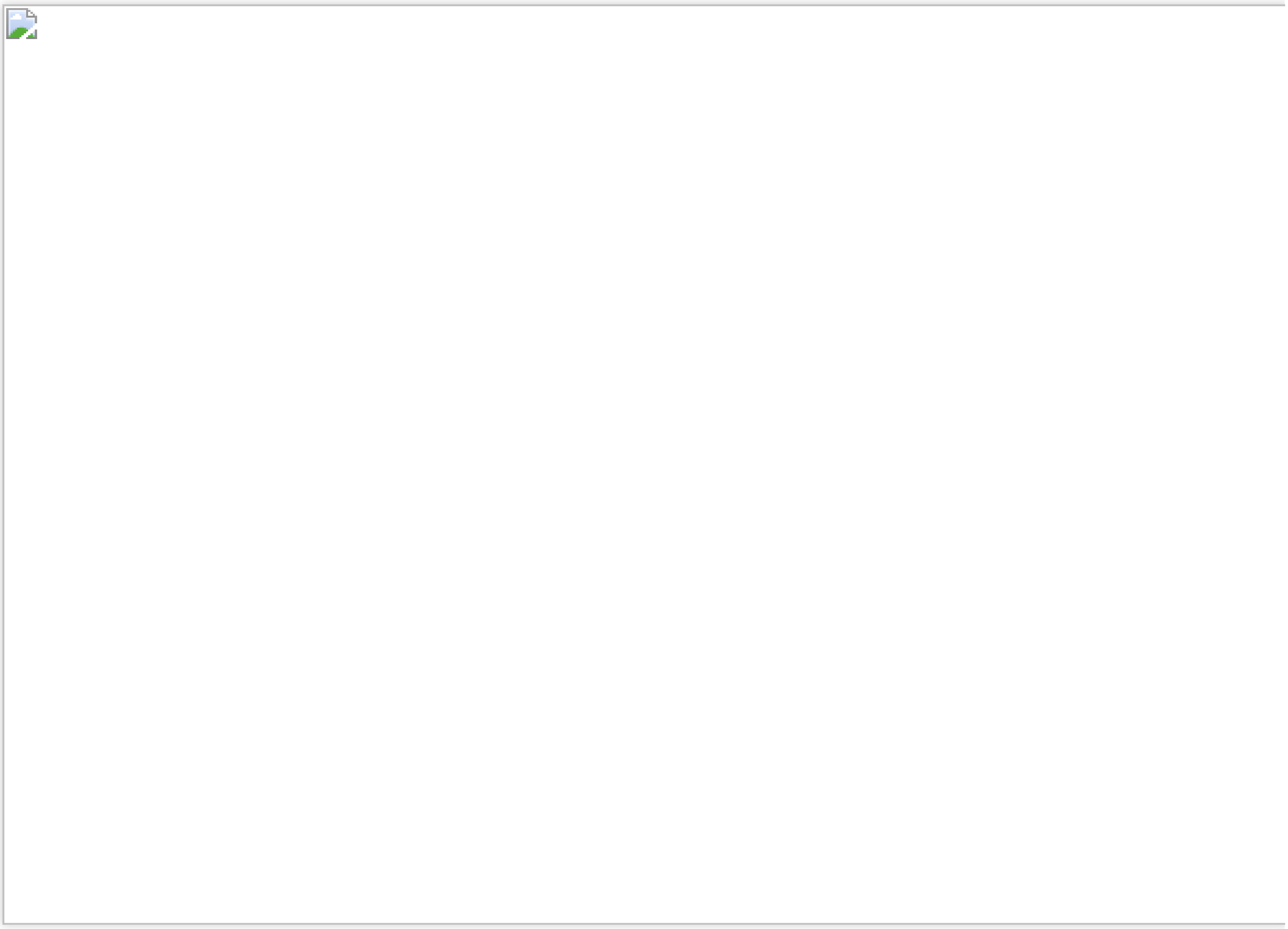
System CPU usage



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	

Compliance

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TRTC meets the compliance requirements of multiple countries and industries. We are dedicated to ensuring the **security, compliance, availability, confidentiality**, and **privacy** of our services. We offer support that helps you and your customers **meet different regulatory requirements so you can reduce auditing costs and improve your auditing and management efficiency.**

TRTC has passed SOC 1, SOC 2, and SOC 3 audits, meets the requirements of China Cybersecurity Law MLPS 2.0, and is certified to ISO 9001, ISO 20000, ISO 27001, ISO 27017, ISO 27018, ISO 27701, ISO 29151, CSA STAR, NIST CSF, BS 10012, and K-ISMS.