

Tencent Real-Time Communication

SDK Download

Product Documentation



Copyright Notice

©2013-2022 Tencent Cloud. All rights reserved.

Copyright in this document is exclusively owned by Tencent Cloud. You must not reproduce, modify, copy or distribute in any way, in whole or in part, the contents of this document without Tencent Cloud's the prior written consent.

Trademark Notice



All trademarks associated with Tencent Cloud and its services are owned by Tencent Cloud Computing (Beijing) Company Limited and its affiliated companies. Trademarks of third parties referred to in this document are owned by their respective proprietors.

Service Statement

This document is intended to provide users with general information about Tencent Cloud's products and services only and does not form part of Tencent Cloud's terms and conditions. Tencent Cloud's products or services are subject to change. Specific products and services and the standards applicable to them are exclusively provided for in Tencent Cloud's applicable terms and conditions.

Contents

SDK Download

SDK Download

Release Notes (App)

Release Notes (Web)

Release Notes (Electron)

SDK Download

SDK Download

Last updated : 2022-09-23 18:02:58

SDK Download

TRTC is a set of low-latency and high-quality audio/video communication services provided by Tencent Cloud. It offers stable and reliable audio/video transmission capabilities at a low cost. TRTC services are implemented via a global transmission network and an SDK. You can find below different editions of the TRTC SDK, which cover mainstream platforms and software frameworks.

Web

Android

iOS

Windows

macOS

Cross-platform

Edition Comparison

Module	Feature	TRTC Lite	TRTC Professional
--------	---------	-----------	-------------------

Video call	One-to-one call	✓	✓
	Group conference	✓	✓
Co-anchoring	Same-room communication	✓	✓
	Cross-room communication	✓	✓
Basic beauty filters	Skin brightening/smoothing	✓	✓
	Color filters	✓	✓
Publishing	Camera	✓	✓
	Screen	✓	✓
Playing	RTMP	-	✓
	HTTP-FLV	✓	✓
	HLS (M3U8)	-	✓
	Live Event Broadcasting (WebRTC)	-	✓
Video on demand	MP4	-	✓
	HLS (M3U8)	-	✓
	DRM	-	✓
Short video making	Recording/Shooting	-	✓
	Clipping/Splicing	-	✓
	TikTok-like special effects	-	✓
	Video upload	-	✓
Size	Android	ARMv7: 3.97 MB ARM64: 4.33 MB	ARMv7: 9.15 MB ARM64: 10.4 MB
	iOS	ARM64: 3.15 MB	N/A

notice

The Windows and macOS SDKs integrate TRTC, TXLivePlayer, and TXVodPlayer, but do not offer short video capabilities. They do not come in lite and professional editions.

Release Notes (App)

Last updated : 2022-09-21 16:46:12

Version 10.3 Released on July 8, 2022

New features:

- Windows: Added support for recording live streaming sessions and audio/video calls to local storage. For details, see the description of `ITXLiteAVLocalRecord` .
- Windows & macOS: Added a parameter to `startMicDeviceTest` , which allows you to specify whether to play the audio captured during mic testing. For details, see the description of `startMicDeviceTest` .

Improvement:

All platforms: Improved audio quality in the `Music` mode.

Bug fixing:

- All platforms: Fixed occasional errors for the user list callback.
- Windows: Fixed the issue where videos sometimes freeze during playback.
- Windows: Fixed occasional video playback failure.
- Windows: Fixed the echo issue for custom audio capturing.

Version 10.2 Released on June 23, 2022

New features:

- All platforms: Launched a new API for stream mixing and relaying, which offers more powerful features and greater flexibility. For details, see the description of `startPublishMediaStream` .
- All platforms: Added support for 3D spatial audio. For details, see the description of `enable3DSpatialAudioEffect` .
- All platforms: Added support for voice activity detection. This feature works even when local audio is muted (`muteLoalAudio`) or the capturing volume is set to zero (`setAudioCaptureVolume`). It allows you to remind users when they are talking but have not turned their mics on. For details, see the description of `enableAudioVolumeEvaluation` .
- All platforms: Added support for checking a user's permission when they switch roles. For details, see the description of `switchRole(TRTCRoleType role, const char* privateMapKey)` .
- iOS & macOS: The C++ API for custom pre-processing supported using textures for video processing.

Improvement:

- Android: Optimized in-ear monitoring, reducing latency.

- Android: Optimized audio capturing, fixing the issue of noise on some devices.
- iOS: Optimized the processing of upstream video data, reducing CPU and GPU usage.
- Windows & macOS: Improved encoding for screen sharing. The height and width of the output video are no longer limited by the window size.
- Windows: Reduced memory fragmentation and performance overhead.

Bug fixing:

- All platforms: Fixed the issue where data sometimes fails to be published after a switch to a different network type.
- iOS: Fixed the issue of noise in recording files saved locally on some iOS 14 systems.

Version 10.1 Released on June 6, 2022**New features:**

- All platforms: Supported smooth role switch so that audio and video playback will not be interrupted momentarily during role switch.
- iOS: Supported stereo audio capturing.
- Android: Added support for capturing system audio (`startSystemAudioLoopback`) on Android 10 and later.

Improvement:

- All platforms: Optimized the echo cancellation capability in the music scenario to make the sound quality more natural.
- All platforms: Optimized the sound quality and startup effect when the role is switched and `muteLocalAudio` is called.
- All platforms: Optimized the callback for bandwidth prediction `onSpeedTest` .
- iOS: Optimized memory management to avoid heap memory issues.
- Android: Reduced the delay of in-ear monitoring on certain phones.
- Windows: Optimized the rendering of downstream video data.
- Windows: Optimized the stereo capturing logic to effectively avoid the problem of echo.

Bug fixing:

- All platforms: Fixed the `reason` exception of the callback for room exit (`onExitRoom`).
- All platforms: Fixed the issue of black screen when the timestamps are equal during custom video sending.
- All platforms: Fixed the crash issue when `muteLocalAudio` and `startLocalAudio` are called successively.
- All platforms: Fixed the issue where 3A is turned on automatically when custom audio capturing is enabled.
- All platforms: Fixed the occasional issue of noise in custom audio rendering.
- iOS: Fixed a memory leak when the log path is set midway (`setLogDirPath`) and the sandbox changes.

- iOS & macOS: Fixed the crash in the continuous background music playback scenario when the system audio service is abnormal.
- Android: Fixed the occasional issue where Bluetooth headsets constantly reconnect.
- Android: Fixed the occasional no audio issue on certain phones.
- Android: Fixed the crash caused by repeatedly plugging and unplugging the earphone on certain phones such as Redmi.
- Windows & iOS: Fixed an issue that causes screencapturing failure.
- Windows: Fixed the issue where, after the mirror mode is enabled for the VOD player, the SDK always crashes when the VOD player is closed.
- Windows: Fixed the issue where `generateCustomPts` is not used in PTS streaming so multiple streams may cause PTS to roll back.
- Windows: Fixed the issue where the SDK occasionally crashes when video is disabled and an image is displayed.

Version 10.0 Released on May 17, 2022

Improvement:

- All platforms: Improved the speed of callback for the anchor's room entry and exit ([onRemoteUserEnterRoom](#) / [onRemoteUserLeaveRoom](#)).
- Windows: Optimized the performance of screen sharing so that the performance is doubled when the filter window is not set.

Bug fixing:

- iOS & macOS: Fixed the occasional [onComplete](#) callback error when starting to play back the background music.
- Android: Fixed the crash caused by the network module.
- All platforms: Fixed the issue of abnormal SEI sending.

Version 9.9 Released on May 6, 2022

Improvement:

- Windows: Optimized the video linkage to reduce performance overheads.
- Windows: Optimized preprocessing for `Systemloopback` capturing to preserve the effect of two sound channels.
- macOS: Reduced crackling caused by a high capturing volume, improving the sound experience.
- macOS: Improved the quality of screen sharing (substream).
- Android: Optimized the capturing delay and improved the in-ear monitoring experience.

Bug fixing:

- Android: Fixed the issue where room IDs do not support numbers above 2.1 billion.

Version 9.8 Released on April 21, 2022

New features:

- Windows: Added APIs for audio effects such as heavy metal and little girl. For details, see `ITXAudioEffectManager.setVoiceChangerType` .
- Windows: Added support for showing an image when local video is paused.

Improvement:

All platforms: Improved the SDK performance in video scenarios.

Bug fixing:

- macOS: Fixed the issue where driver installation fails when the audio of the built-in sound card is recorded.
- All platforms: Fixed custom rendering failure when the local screen is shared (via the substream).

Version 9.7 Released on April 6, 2022

Improvement:

- iOS & Android: Improved audio quality in the music mode.

Note:

You can use the [TRTCCloud.startLocalAudio \(TRTCAudioQualityMusic\)](#) API to enable the music mode on all platforms.

- Windows: Improved audio quality and reduced audio loss in the music mode.
- Windows: Improved compatibility with high-end sound cards, boosting audio quality.
- Windows: Optimized audio mixing with third-party processes for more scenarios.

Bug fixing:

- All platforms: Fixed occasional issue of blurry screen during CDN playback.
- iOS & Android: Fixed failure to switch between the receiver and speaker when playing live streams.
- iOS & Android: Fixed the issue where, after an API is called to enable the music mode, audio quality doesn't meet expectations.
- iOS: Fixed occasional memory leaks during software encoding.
- iOS: Fixed the occasional issue where the first frame callback for local video rendering is not received.
- Windows: Fixed the occasional issue where the SDK crashes when the mouse cursor is captured during screen sharing.

- Windows: Fixed the issue where the speaker fails to work properly in the music mode.
- Windows: Fixed failure to turn on some cameras when the `startCameraDeviceTest` API is called.

Version 9.6 Released on March 24, 2022

*Major changes:

- All platforms: Enhanced third-party library compliance with regulations inside and outside the Chinese mainland.
- All platforms: Reduced the SDK storage footprint. For details, see the table below:

Platform	Before	After
Android	ARMv7: 6.95 MB ARM64: 7.94 MB	ARMv7: 4.32 MB ARM64: 4.85 MB
iOS	ARM64: 3.23 MB	ARM64: 3.15 MB
Windows	Win32: 21.3 MB Win64: 26.9 MB	Win32: 15.0 MB Win64: 17.2 MB
macOS	x86_64: 18.1 MB	x86_64: 15.8 MB

Bug fixing:

All platforms: Fixed known issues, improving stability.

Improvement:

- iOS: Fixed occasional overexposure when fill lights are used.
- macOS: Optimized texture uploading.
- Android: Improved pre-processing methods for beauty filters and other features, fixing capturing stutter on low-end devices.
- Windows: Updated the live streaming component from V1 to V2 APIs, improving its stability.
- Windows: Improved compatibility with the GPUs of low-end devices.

Version 9.5 Released on January 11, 2022

Bug fixing:

- All platforms: Fixed the issue where, when users make API requests in a certain order to enable custom video rendering, a black screen is displayed.
- Windows: Fixed the issue where only a portion of the specified part of the screen is captured during screen sharing.
- iOS: Fixed the issue where, if a user calls `muteLocalVideo` before leaving a room and then enters a new room, publishing is still disabled.
- iOS: Fixed failure to set a background image for stream mixing.

Improvement:

- All platforms: Improved the smoothness of calls under poor network conditions.
- Windows: Fixed occasional failure to start cameras and the issue where some cameras fail to capture videos at the specified frame rate, improving browser compatibility.
- iOS: improved compatibility with rendering components such as Cocos2d.
- Android: Fixed the issue where, after an anchor turns the camera off and on again, at the player end, before the anchor's video is played as expected, the last frame before the camera is turned off is shown first.

Version 9.4 Released on December 8, 2021**New features:**

- All platforms: supported highlighting the speaking user, which is useful in large-scale audio co-anchoring scenarios. It enables users to focus on the audio of whoever is speaking when there are multiple speakers in the room. You can call the [setRemoteAudioParallelParams](#) API to set this feature.
- macOS: supported dual channels for the system audio capturing API [startSystemAudioLoopback](#).
- iOS: supported background music files in 24-bit WAV format.
- Android & iOS: This version complies with China's privacy and security regulations and has been tested by multiple Tencent products.

Bug fixing:

- All platforms: fixed the issue where room switching fails if [switchRoom](#) is called frequently.
 - iOS: fixed the issue where [setVideoEncoderRotation](#) does not work during in-app screen sharing ([startScreenCaptureInApp](#)).
- iOS: fixed the occasional issue of rising memory usage during screen sharing ([startScreenCaptureByReplaykit](#)).

Improvement:

- All platforms: sped up room entry, reducing the fluctuation in room entry speed.
- macOS: fixed the issue of high CPU and memory usage when mouse cursor is captured during screen sharing.
- Android: made the capturing resolution in line with the resolution of the screen during screen sharing to avoid black bars.
- Android: improved the compatibility of the hardware video decoder, fixing the issue of black bars due to change of playback resolution on some devices.
- Windows: optimized the audio gain control algorithm, fixing the issue of marked noise due to high audio gain on some devices.

Version 9.3 Released on November 3, 2021**Bug fixing:**

- All platforms: fixed failure to obtain `point2PointDelay` (value is 0).
- All platforms: fixed occasional parsing failure, which causes SEI messages to be lost.
- macOS: fixed the issue where the camera does not output frames on macOS 12 beta.
- iOS & macOS: fixed the issue where no images are displayed when `startRemoteView` is called earlier.
- Windows: fixed the issue of aliasing when video is encoded in portrait mode and beauty filters are enabled.
- Windows: fixed the issue where, when third-party beauty filters are used, no callback for custom rendering is returned after resolution change.

Improvement:

- All platforms: improved instant streaming performance under poor network conditions.
- All platforms: optimized the QoS control policy under poor network conditions, ensuring smoother communication.
- All platforms: improved the speed test to support testing the current bandwidth.
- All platforms: improved support for TCP for better adaptability to different network environments.

Version 9.2 Released on September 23, 2021**New features:**

- Android & iOS: supported SOCKS5 proxies.
- Windows: enabled adaptive echo cancellation for the `TRTCAudioQualityMusic` mode to automatically balance between audio quality and echo cancellation strength.
- All platforms: allowed audio pitch setting.

Bug fixing:

- Windows: fixed the issue where some cameras do not output data on Windows installed on Mac.
- Android: fixed the issue where there is no upstream audio after switch between CDN live streaming and TRTC.
- iOS: fixed the issue where, when a screen is shared from web, users on iOS see blurry video if they enable custom rendering.

Improvement:

- Android: fixed the issue where the “Application Not Responding” error occurs during hardware decoding.
- Android: fixed the compatibility issue for the rotation of local camera preview.
- Android: improved instant streaming performance.
- Android & iOS: optimized the 3A policy for the duet mode.
- Windows: improved the AGC algorithm, reducing cases of excessively low or high volume.
- All platforms: optimized the jitter control algorithm under poor network conditions, enabling smoother video playback.

Version 9.1 Released on September 4, 2021

New features:

- All platforms: supported using a C++ API to set the format of called back audio frames.
- Windows: supported streaming VOD files in AC3 format.
- Windows: supported getting the resolutions supported by a camera. For details, please see [ITXDeviceCollection.getDeviceProperties](#).
- Windows: supported NVIDIA, Intel, and AMD hardware decoding.
- macOS: supported recording local media.

Bug fixing:

- All platforms: fixed occasional failure to enter a room.
- macOS: fixed the issue where, during screen sharing, the preview flickers when the resolution is changed.
- Android: fixed display error of the substream video after switching from a sub-room to the main room.
- Android: fixed the occasional issue where the frame rate setting does not take effect in some scenarios.
- Windows: fixed failure to pull streams after audience switch to CDN playback.
- Windows: fixed the issue where the image disappears when VOD files in certain formats are streamed.

Improvement:

- All platforms: improved experience under poor network conditions.
- Android: improved audio status management during room exit.
- Android: improved the logic of recovery in the case of audio capturing failure, to increase the success rate of audio capturing.
- Android: fixed video overexposure under certain conditions.

Version 9.0 Released on August 6, 2021

New features:

- iOS: allowed setting the capturing volume of system audio. For details, please see [setSystemAudioLoopbackVolume](#).
- All platforms: allowed setting the volume of custom audio tracks. For details, please see [setMixExternalAudioVolume](#).
- All platforms: separated audio and video packet loss in the status callback. For details, please see [TRTCRemoteStatistics](#).

Improvement:

- All platforms: optimized the subscription process to improve instant streaming performance for manual subscription.
- All platforms: fixed the issue of repeated `onExitRoom` callback in some scenarios.

Bug fixing:

- Android: fixed the issue where bitrate and frame rate settings during custom capturing do not take effect.
- iOS: fixed failure to publish streams if users enable the screen sharing substream first and then turn the camera on.
- iOS: fixed blurriness of recorded local video.
- iOS: fixed several stability issues.
- Windows: fixed the frame rate exception for the capturing of screen sharing images.
- Windows: fixed the issue where, after the sharing source is changed during screen sharing, audience see a frame of the old source before the new source is played.

Version 8.9 Released on July 15, 2021

New features

- Android: supported specifying external GL contexts for custom capturing, allowing more flexible use of OpenGL contexts.
- Windows: allowed specifying the speaker for system audio capturing ([startSystemAudioLoopback](#)).
- Windows: supported NVIDIA hardware encoding, improving stream publishing performance.
- All platforms: supported cloud proxies, which are a secure and easy way to access TRTC from inside a corporate firewall.
- All platforms: added the stream type parameter to the APIs [muteLocalVideo](#) and [muteRemoteVideoStream](#).
- All platforms: added the gateway RTT parameter `gatewayRtt` to the status callback [onStatistics](#), which indicates the quality of network between users and their Wi-Fi routers.
- All platforms: supported recording audio into more formats using the [startAudioRecording](#) API.

Quality improvement

- All platforms: fixed shaky audio in some scenarios.
- Android: improved instant streaming performance.
- Android: upgraded the audio pre-processing algorithm for clearer audio in calls.

Bug fixing

- Windows: fixed the issue of echo in recorded local video if VODPlayer is used to stream VOD files.
- Windows: fixed crash when the window filtering feature is enabled during screen sharing on high DPI displays.
- iOS: fixed the issue where the landscape mode does not take effect for system-level screen sharing via the substream.

- iOS: fixed the issue of memory leak when custom rendering is enabled only for remote videos and the RGBA format is used.
- All platforms: fixed occasional failure to enter a room.

Version 8.8 Released on June 21, 2021

New features

Android & macOS & iOS: allowed playing audio via peripheral devices. For details, please see the API [enableCustomAudioRendering](#).

Quality improvement

- All platforms: made it easier to use `mixExternalAudioFrame`. You no longer need to call the API at a regular interval.
- macOS: reduced the CPU usage of screen sharing when mouse cursor capturing is enabled.
- Windows: made AGC faster and more timely for better results.
- Windows: reduced the performance overhead of screen sharing when the window filtering feature is enabled.

Bug fixing

- iOS: fixed the issue where, when a local audio file in AAC format is played, the total length is inaccurate.
- Android: fixed the issue of audio stutter after the SDK is switched to the background on some devices.

Version 8.7 Released on May 25, 2021

New features

- All platforms: supported anomaly detection for peripheral audio devices. After registering `onStatistics`, you can detect in real time when there is no audio for a long time and when audio cracks or is interrupted via the `audioCaptureState` field in [TRTCLocalStatistics](#).
- Windows: supported RGBA video data for custom capturing.

Quality improvement

- All platforms: optimized background music management to release memory resources in a timely manner.
- All platforms: ensured that audience receive the `onUserVideoAvailable(false)` notification in a timely manner after stream publishing is paused because the application is switched to the background.
- macOS: reduced the CPU and memory usage of mouse cursor capturing during screen sharing.

Bug fixing

- Android: fixed the issue where `setRemoteViewFillMode` doesn't work on some devices.
- iOS & macOS: fixed the memory release issue of custom beauty filters after they are disabled.

Version 8.6 Released on May 8, 2021

- All platforms: optimized the QoS control algorithm, enhancing audio/video transmission quality.
- All platforms: improved audio playback smoothness when users switch between the anchor and audience roles.
- iOS & macOS & Windows: optimized the audio processing module, improving audio quality in the speech and default modes.
- iOS & macOS: improved the adaptability of custom audio capturing to situations of high CPU usage.
- iOS & Android: supported publishing screen recording data via the substream, as in SDKs for desktop platforms.
- macOS: added native support for Apple M1.
- Windows: optimized the memory allocation logic to enhance stability.

Version 8.5 Released on March 24, 2021

New features

- macOS: optimized the screen sharing feature. You can now share other windows along with the target window. For details, see the API [addIncludedShareWindow](#).
- All platforms: supported publishing VOD content. You can now bind [TXVodPlayer](#) with `TRTCCloud` and publish the content played by VOD via TRTC's substream.
- All platforms: supported custom capturing of substream data. For details, see the API [sendCustomVideoData](#).
- All platforms: supported custom audio mixing. You can feed a local audio track into the SDK's audio processing. The SDK will mix the two tracks before publishing. For details, see the API [mixExternalAudioFrame](#).
- All platforms: supported mixing only video streams, allowing for more flexible stream mixing control.

Quality improvement

- macOS: enabled the `startSystemAudioLoopback` API to support dual sound channels.
- Windows: supported automatic switch to the slideshow window when a slideshow is selected for screen sharing.
- All platforms: included end-to-end latency in the information returned via the status callback.

Bug fixing

- iOS: fixed occasional crash when images are rendered using OpenGL in the background.
- iOS: fixed playback failure when shared images are static.

Version 8.4 Released on February 8, 2021

New features

- macOS: supported capturing system audio, i.e., the system loopback feature that is enabled on Windows. The feature allows the SDK to capture system audio so that anchors can stream local audio or video files to other users.

- macOS: supported local preview for screen sharing. You can now display screen sharing preview in a small window.
- Windows: supported setting the volume of the current process. You can now use `setApplicationPlayVolume` to set the volume of the volume mixer.
- All platforms: supported local recording. An anchor can now record local audio and video into an MP4 file during streaming. For details, see [startLocalRecording](#).

Quality improvement

- All platforms: improved audio quality in the [music](#) mode, which makes it more suitable for Clubhouse-like audio streaming scenarios.
- All platforms: improved the adaptability to poor network conditions. Smooth audio and video can be delivered even when the packet loss rate reaches 70%.
- Windows: improved audio quality in some streaming scenarios by significantly reducing audio damage.
- Windows: improved performance by 20%-30% in some scenarios.

Bug fixing

- Windows: fixed the issue where the SDK crashes when the desktop is shared on Windows Server 2019 Datacenter x64.
- Windows: fixed the issue where screen sharing sometimes ends unexpectedly when the target window is resized during screen sharing.
- Windows: fixed image capturing failure with some cameras.
- iOS: fixed the issue where `snapvideoshot` causes stuttering with `CAAnimation`.
- iOS & macOS: fixed the black screen issue when the same view is used to display camera and screen sharing images in turn.
- iOS: fixed the blurry screen issue on iPhone 6S when a third-party beauty filter component is used.
- iOS: fixed the issue where, in cases where TRTC and VOD are used at the same time, the SDK occasionally crashes after VOD is stopped.
- Android: fixed the issue where audio is played via the speaker after a user using Bluetooth earphone rejects an incoming call.

Version 8.3 Released on January 15, 2021

New features

Optimized the business logic of custom capturing:

- iOS & Android & macOS: optimized the audio module to ensure acoustic echo cancellation (AEC) and active noise suppression (ANS) effects when you use [enableCustomAudioCapture](#) to capture audio data and send it to the SDK for processing.

- iOS & Android: if you need to add your own audio effects and audio processing logic in addition to those of the TRTC SDK, we recommend you use version 8.3, with which you can use [setCapturedRawAudioFrameDelegateFormat](#) and other APIs to set what to include in the audio data callback, for example, the audio sample rate, the number of sound channels, and the number of samples, so that you can process audio data in your preferred format.
- All platforms: if you collect video data by yourself and use the audio module of the TRTC SDK at the same time, lip-sync errors may occur. This is because the SDK has its own timeline control logic. To solve this problem, we have provided the [generateCustomPTS](#) API. When a video image frame is captured, call this API and record the PTS (timestamp), and provide the timestamp when you call [sendCustomVideoData](#).
- Windows: supported SOCKS5 proxy addresses for domain names.

Bug fixing

- All platforms: fixed occasional lip-sync errors for recorded content due to timestamp exceptions in audio data.
- Windows: improved the compatibility of window sharing with high DPI displays.
- Windows: added minimized windows to the shareable window list. The thumbnails of minimized windows are their application icons.
- Windows: fixed the issue of unnecessary CPU usage by DXGI after the SDK is started.
- iOS: fixed the ANR error caused by manual focus setting.
- iOS: fixed occasional failure to switch between the front and rear cameras.
- iOS: fixed VODPlayer crash when video is played back in slow motion.
- iOS: fixed the issue where audio is occasionally played via the speaker after room entry.
- iOS & Android: optimized the AEC and ANS effects and supported reverb effects for in-ear monitoring.
- Android: fixed occasional green or blurry screen during hardware decoding.
- macOS: fixed the issue where, during screen sharing with the highlighting feature enabled, the highlighted borders of the shared window flash when the window is moved near the edge of the screen.
- macOS: fixed black screen when the view rendered is moved.

Version 8.2 Released on December 23, 2020

New features

- iOS & Android: supported callback of a combination of locally captured audio and all played back audio, making local recording easier.
- Android: the video rendering component `TXCloudVideoView` supported using `TextureView` for local rendering through the calling of the `addVideoView(new TextureView(getApplicationContext()))` API.
- Android: supported RGBA video data for custom rendering.
- Windows: supported taking screenshots of locally captured video and played back remote video. For details, see [ITRTCCloud.snapshotVideo](#).

- Windows: supported using `addExcludedShareWindow` and `addIncludedShareWindow` to exclude or include windows you specify, increasing the flexibility of screen sharing.
- macOS & iOS: supported calling `TRTCCloud.snapshotVideo` to take screenshots of video in the custom rendering mode.

Quality improvement

- Android: improved encoding quality for live streaming, enabling clearer video images.
- Windows: improved the AEC algorithm.

Bug fixing

- iOS: fixed occasional audio playback errors when VODPlayer and TRTC are used at the same time.
- Android: fixed black screen when custom beauty filters are used.
- Windows: fixed occasional failure to exit the current process.

Version 8.1 Released on December 3, 2020

New features

- All platforms: added statistics on remote video stuttering to `onStatistics` .
- All platforms: supported using the volume adjustment API `setAudioPlayoutVolume` (100-150) to enable audio gain.
- iOS & Android: added the `setLocalVideoProcessListener` API to better support the integration of third-party beauty filters.
- C#: upgraded to the latest APIs.

Quality improvement

- All platforms: optimized the audio processing algorithm when earphones are used to deliver better audio quality.
- Android: optimized the audio pre-processing algorithm to reduce the impact of AEC, ANS, and AGC on audio quality.

Bug fixing

- iOS: fixed occasional crash when the application is force killed.
- Android: fixed the issue where beauty filters do not produce desired results when the frame rate of captured video is high.
- Windows: fixed occasional crash during screen sharing when high DPI displays are used.

Version 8.0 Released on November 13, 2020

Added

- All platforms: added cross-platform C++ APIs. For more information, see `cpp_interface/ITRTCCloud.h`.
- All platforms: supported string-type room IDs. For more information, see `TRTCParams.strRoomId`.
- All platforms: added the device management class `TXDeviceManager`.
- All platforms: added the `TRTCCloud.switchRoom` API to allow room switching with capturing uninterrupted.
- All platforms: added the `TRTCCloud.startRemoteView` API to start rendering remote video images.
- All platforms: added the `TRTCCloud.stopRemoteView` API to stop rendering remote video images.
- All platforms: added the `TRTCCloud.getDeviceManager` API to get the device management class.
- All platforms: added the `TRTCCloud.startLocalAudio` API to enable local audio capturing and upstream data transfer.
- All platforms: added the `TRTCCloud.setRemoteRenderParams` API to configure rendering for remote images.
- All platforms: added the `TRTCCloud.setLocalRenderParams` API to configure rendering for local images.

Optimization

- Android: optimized the logic for switching between software and hardware decoding.
- Windows: improved audio quality and AEC for system loopback.
- Windows: optimized the audio device selection logic to reduce cases of no audio.
- Windows: reduced audio loss in double-talk scenarios.
- All platforms: optimized instant streaming for role switching in the manual subscription mode.
- All platforms: optimized the audio receiving logic, improving audio quality.
- All platforms: improved the reliability of `sendCustomCmdMsg`.

Bug fixing

- iOS: fixed the issue where the calling of `muteLocalVideo` suspends local video rendering.
- iOS: fixed the issue where the application occasionally freezes when a component is called during foreground-background switch.
- iOS: fixed intermittent in-ear monitoring audio when audio effects are enabled.
- Android: fixed the issue where audio effects played in the call volume mode do not stop when there is an incoming call.
- Android: fixed occasional failure to enable audio capturing.
- Windows: fixed occasional black screen during local video rendering.
- Windows: fixed occasional crash when users exit the app.
- Windows: improved support for Bluetooth earphones and fixed the no audio issue.
- Windows: fixed the focus stealing problem that occurs when screen sharing stops.
- All platforms: fixed failure to collect statistics on packet loss rate for status callback.

Version 7.9 Released on October 27, 2020

Added

- macOS: supported filtering out selected windows from screen sharing. Users can exclude windows they do not want to share, better ensuring privacy.
- Windows: supported configuring the border color and width of the "Sharing" message box during screen sharing.
- Windows: supported the high performance mode during desktop sharing.
- All platforms: supported custom encryption, allowing users to process encoded audio/video data using an exposed C API.
- All platforms: added audio stuttering information `audioTotalBlockTime` and `audioBlockRate` to `TRTCRemoteStatistics`.

Optimization

- iOS: shortened the startup time of the audio module, allowing quicker capturing and sending of the first audio frame.
- Windows: optimized the AEC algorithm for system audio loopback.
- Windows: allowed users to filter out certain windows from screen sharing to prevent the target window from being covered.
- Android: optimized the in-ear monitoring effect for most Android devices, reducing in-ear monitoring latency to a more acceptable level.
- Android: reduced end-to-end delay in the music mode (specified in `startLocalAudio`).
- All platforms: enhanced audio smoothness when a user switches between the anchor and audience role in the manual subscription mode.
- All platforms: improved audio/video call performance and audio smoothness in poor network conditions.

Bug fixing

- iOS: fixed occasional failure to render video images in certain scenarios.
- iOS: fixed occasional noise when users use earphones in the default mode.
- iOS: fixed known memory leak issues.
- iOS: fixed occasional crash after ReplayKit screen recording ends.
- iOS: fixed compilation problems on simulators.
- Android: fixed occasional lip-sync errors on certain phones after the application remains in the background for a long time.
- Android: fixed the issue where the mic is not released after the application is switched to the background.
- Android: fixed the issue where certain OpenGL resources in the SDK are not released in time.
- Windows: fixed occasional noise in some scenarios.
- All platforms: fixed occasional crash, improving the SDK's performance stability.

Version 7.8 Released on September 29, 2020

Added

- macOS: added the callback of system volume change. For details, please see [TRTCCloudDelegate.onAudioDevicePlayoutVolumeChanged](#).
- Windows: supported specifying content for screen sharing across screens.
- Windows: supported filtering out specified windows from screen sharing to prevent the target window from being covered. For more information, please see [TRTCCloud.addExcludedShareWindow](#) and [TRTCCloud.removeExcludedShareWindow](#).
- Windows: added the callback of system volume change.

Optimization

- iOS: allowed using VODPlayer and TRTC at the same time with AEC enabled.
- iOS & macOS: supported pushing a specified image when stream pushing pauses. For more information, please see [TRTCCloud.setVideoMutelImage](#).
- Android: supported pushing a specified image when stream pushing pauses. For more information, please see [TRTCCloud.setVideoMutelImage](#).
- Android: optimized the audio routing policy to make sure that audio is always played back via earphones when earphones are connected.
- Android: allowed low-delay capturing and playback in certain systems, reducing call delay.
- Android: supported using VODPlayer and TRTC at the same time with AEC enabled.
- Windows: made the SDK compatible with the virtual webcam e2eSoft VCam.
- Windows: allowed calling `startLocalPreview` and `startCameraDeviceTest` at the same time.
- Windows: allowed publishing screen sharing images via the primary stream and at the same time calling `startLocalPreview` to enable local preview.
- Windows: fixed long audio delay caused by the playback buffer of the SDK.
- Windows: optimized the audio enablement logic to prevent mic occupation in the playback-only mode.

Bug fixing

- iOS: fixed low playback volume on iPhone SE.
- iOS: fixed crash when a sub-instance (`TRTCCloud.createSubCloud`) calls `muteRemoteAudio` .
- iOS: fixed occasional crash during rendering.
- iOS: fixed the issue where video rendering on certain iPad devices occasionally causes the main thread to crash during foreground/background switching.
- iOS: fixed known memory leak issues.
- iOS: fixed the issue where iOS 14 prompts "the app would like to find and connect to devices on your local network".

- macOS: fixed the issue where `getCurrentCameraDevice` returns `nil`.
- macOS: fixed the issue where certain USB cameras cannot be turned on.
- macOS: fixed crash when the area of shared content is set to `0`.
- Android: fixed crash on Android 5.0 when the `READ_PHONE_STATE` permission is not granted.
- Android: fixed audio capturing and playback exceptions after Bluetooth earphones are disconnected and connected again.
- Android: fixed known crash issues.
- Windows: fixed crash on 64-bit Windows when screen sharing is enabled and disabled multiple times.
- Windows: fixed crash on certain systems when OpenGL is used.

Version 7.7 Released on September 8, 2020

Optimization

- All platforms: improved instant streaming performance of the substream (screen sharing images).
- iOS: optimized the internal thread model to improve stability when 30 or more channels of audio/video are played back at the same time.
- iOS & Android: improved the performance of the audio module and reduced the capturing delay of the first audio frame.
- iOS & Android: improved volume and audio quality when VODPlayer and TRTC are used at the same time.
- iOS & Android: supported files in WAV format for audio effects and background music.
- Windows: fixed the issue of high CPU utilization when some low-end cameras are used.
- Windows: optimized the compatibility with multiple USB cameras and mics to make it easier to turn on such devices.
- Windows: optimized the selection policy of cameras and mics to avoid audio/video capturing exceptions caused by the connection/disconnection of cameras and mics.

Bug fixing

- All platforms: fixed occasional playback exceptions when the `muteLocalVideo` and `muteLocalAudio` APIs are called in poor network conditions.
- iOS: fixed occasional failure to play audio effects on earlier generations of iPhone or iPad devices.
- iOS: fixed distorted screen sharing images on iPad Pro.
- iOS: fixed the issue where the application keeps requesting screen recording permission after the user denies it.
- Windows: fixed the issue where `onExitRoom` fails after laptops or desktops remain in sleep mode for a long time.
- Windows: fixed echo after system audio capturing is enabled via the calling of `startSystemAudioLoopback` in the music mode.
- Windows: fixed the issue where no audio is played sometimes when a user uses `enterRoom` and `exitRoom` to enter and leave the room in a short period of time.
- Windows: fixed project compilation problems with Visual Studio 2010.

- Windows: fixed the issue where the `onUserVideoAvailable` event callback is returned multiple times in the manual subscription mode (`setDefaultStreamRecvMode(false, false)`).

Version 7.6 Released on August 21, 2020

Added

- Windows: added the `updateLocalView` and `updateRemoteView` APIs to improve user experience in adjusting HWND rendering windows in real time.
- Windows: added the `getCurrentMicDeviceMute` API to get whether the PC is muted.
- Windows: added the `setCurrentMicDeviceMute` API to turn on global mute for the PC.
- macOS: added the `updateLocalView` and `updateRemoteView` APIs to optimize user experience in adjusting the view rendering area in real time.
- macOS: added the `getCurrentMicDeviceMute` API to get whether the PC is muted.
- macOS: added the `setCurrentMicDeviceMute` API to turn on global mute for the PC.
- iOS: added the `updateLocalView` and `updateRemoteView` APIs to optimize user experience in adjusting the view rendering area in real time.
- iOS: added the `onCapturedRawAudioFrame` callback to `TRTCCloudDelegate`, and changed the names of a number of other callback functions. The names used now are `onLocalProcessedAudioFrame`, `onRemoteUserAudioFrame`, and `onMixedPlayAudioFrame`.
- Android: added the `onCapturedRawAudioFrame` callback to `TRTCCloudListener`, and changed the names of a number of other callback functions. The names used now are `onLocalProcessedAudioFrame`, `onRemoteUserAudioFrame`, and `onMixedPlayAudioFrame`.

Optimization

- All platforms: optimized the protocol policy of `enterRoom` to improve the speed and success rate of room entry.
- All platforms: fixed reduced performance and stuttering when a large number of audio channels are subscribed at the same time.
- macOS: supported sharing specified area of a specified window.

Bug fixing

- All platforms: fixed the issue where the SDK does not trigger the `onEnterRoom` callback when users enter the same room without exiting.
- All platforms: fixed a few internal bugs that may cause a black screen.
- All platforms: fixed failure to display screen sharing images when `startRemoteSubStreamView` is called early.
- Windows: fixed known handle and GDI leaks.
- Windows: fixed known crash issues.

- Windows: fixed the issue where cameras and mics are not started automatically after being disconnected and connected again.
- iOS: fixed crash on iOS 10 when certain file paths are passed in the background music API.
- Android: fixed the occasional no audio issue when `enterRoom` and `exitRoom` are called multiple times in a short period of time.
- Android: fixed occasional black screen during the streaming of screen recording.

Version 7.5 Released on July 31, 2020

Added

- Supported dual-stack IPv6 and IPv6-only.
- Allowed playing back streams in multiple rooms. This feature can be used for ultra-small classes.
- Allowed setting a background image for MCU On-Cloud MixTranscoding (for regulatory purposes, the image must be uploaded to the TRTC console first).
- Added two new modes for MCU On-Cloud MixTranscoding: `A+B=>C` and `A+B=>A`.
- Added the `jitterBufferDelay` field, which indicates the playback buffer time, to the real-time status callback API `onStatistics`.

Optimization

- Reduced end-to-end delay for co-anchoring by 40% from that in version 7.4.
- Reduced in-ear monitoring delay on phones and allowed setting voice change and reverb effects for in-ear monitoring.
- Optimized the algorithm for evaluating network jitter at the player end to reduce playback delay.
- Reduced end-to-end delay for co-anchoring in TRTC SDK for Android.
- Reduced in-ear monitoring delay.
- Optimized the issue where playback view switching causes a black screen.

Bug fixing

- Fixed the issue where playback fails after `playBGM` and `pauseBGM` are called successively in a function.
- Fixed the occasional issue where users continue to receive the `onEnterRoom` callback after exiting the room.
- Fixed the issue on certain devices where ultra-low-resolution encoding fails and cannot be recovered.

Version 7.4 Released on June 24, 2020

Optimization

Allowed setting volume for in-ear monitoring.

Bug fixing

- Fixed the issue where the local image flickers during landscape/portrait mode switch on Android.
- Fixed encoding failure when custom videos are published from certain Android phones.
- Fixed occasional crash during audio packet processing.

Version 7.3 Released on June 1, 2020

Added

- Added a new audio effect management API `TXAudioEffectManager` to offer more diverse audio effects while continuing to support the legacy API.
- Added the `minVideoBitrate` option to `setVideoEncoderParam`, the API used to set video encoding parameters. The option is recommended for scenarios with high requirements on video quality.

Optimization

- Supported transient noise reduction, which can be enabled using `setAudioQuality(TRTCAudioQualitySpeech)`.
- Supported asset packages for audio effect files.
- Improved local video clarity.
- Supported custom rendering by texture for the player end, reducing resource consumption.
- Optimized the resolution selection logic for video captured by the camera, enhancing visual experience.
- Optimized echo cancellation.
- Supported 128 Kbps high-quality stereo sound from sender to recipient, which can be set using the `setAudioQuality(TRTCAudioQualityMusic)` API.
- Supported the `SPEECH` audio mode, which provides better ANS capabilities for audio conferencing calls. This mode can be set using the `setAudioQuality(TRTCAudioQualitySpeech)` API.
- Supported playing back multiple tracks of background music at the same time. This feature can be used in karaoke scenarios, where vocals and instrumentals are separate tracks. Loop playback of background music is also supported now.
- Supported calling `muteLocalVideo` before `startLocalPreview` to preview without pushing streams. This can be achieved by calling `startLocalPreview` before `enterRoom` as well.

Bug fixing

- Fixed occasional crash due to OpenGL context error during custom video capturing.
- Fixed the issue where the custom rendering callback is not triggered after `setLocalVideoRenderListener` is called before room entry.
- Fixed the issue where, when a user switches to the front/rear camera in the landscape mode, other users see upside down video of the user.

- Fixed the occasional issue where, if a user calls `startLocalPreview` before room entry, other users see blurry image of the user.
- Fixed occasional crash of the hardware encoder.
- Fixed occasional intermittent audio for local recording.
- Fixed the issue where, when the user publishing streams enters the room again after the pausing of stream pushing (`muteLocalVideo` , `muteLocalAudio`) causes crash or forces the application to close, streams are not played back automatically at the player end.

Version 7.2 Released on April 16, 2020

Added

Supported screen recording on Android, allowing users to stream screen recording on phones.

Optimization

- Optimized call performance on low-end and mid-range Android phones, improving audio experience.
- Optimized visual effect APIs such as filters and green screen and aggregated them into the `TXCBeautyManager` class so that they share a calling method.

Bug fixing

Fixed the occasional issue where the custom stream ID fails to take effect in time during role switching.

Version 7.1 Released on March 27, 2020

Optimization

- Supported static build of projects using the C++ STL library.
- Enabled ANS and AGC by default in the call volume mode to improve audio quality.
- Improved the usability of the preset stream mixing template.
- Enhanced the success rate of stream mixing.

Bug fixing

- Fixed the issue where all audio processing values become zero when AGC is enabled/disabled frequently during room entry.
- Fixed the issue where speed testing slows down the calling of other APIs.
- Fixed the issue where the volume of upstream data doubles and noise is heard after real-time communication is interrupted by an incoming call.
- Fixed the issue where streams are relayed automatically after room entry.

Version 7.0 Released on March 9, 2020

- Optimized the 3A enabling policy.
- Improved the usability of MCU On-Cloud MixTranscoding.
- Enhanced audio smoothness in poor network conditions.
- Fixed memory leaks caused by frequent room entry and exit.

Version 6.9 Released on January 14, 2020

Added

- Supported Android 10.0.
- Added the `snapshotVideo()` API for taking screenshots of local or remote video.
- Added the `pauseAudioEffect` and `resumeAudioEffect` APIs for pausing and resuming an audio effect.
- Added the `setBGMPLayoutVolume` and `setBGMPublishVolume` APIs for setting the local playback volume and publishing volume of background music respectively.
- Added the `setRemoteSubStreamViewRotation` API for adjusting the rotation of played back substream video.
- Added a global volume mode setting API `setSystemVolumeType(TRTCSysVolumeTypeVOIP)`, which can ensure that call volume is used all the time. This is mainly to prevent the switch from Bluetooth earphones to the built-in mic during audio capturing.
- Added the `streamId` attribute to the `TRTCParams` parameter of `enterRoom`, which can be used to set the user's CDN stream ID, making it easier to bind to live streaming CDNs.
- Added the `cloudRecordFileName` attribute to the `TRTCParams` parameter of `enterRoom`, which can be used to set the on-cloud recording filename for a live stream.
- Added the `TRTCAppSceneAudioCall` scenario, which is optimized for audio calls and can be set during the calling of `enterRoom`.
- Added the `TRTCAppSceneVoiceChatRoom` scenario, which is optimized for audio chat rooms and can be set during the calling of `enterRoom`.

Optimization

- Improved the recording feature's tolerance of video interruption, enabling remote recording of more complete video.
- Fixed lip-sync errors during hardware decoding on certain devices.
- Supported capturing 1080p video, allowing PC audience to watch clearer video published from phones.
- Simplified the error codes for room entry.
- Fixed occasional slow streaming.

Bug fixing

- Fixed occasional crash of the HTTP component.
- Fixed the occasional issue where no callback is returned for the completion of audio effect playback.

- Fixed the occasional issue where the system cannot be recovered from room entry failures.

Version 6.8 Released on November 15, 2019

Added

- Added the in-ear monitoring capability.
- Allowed users to disable automatic stream pulling upon room entry.
- Added the `getBeautyManager` API, which aggregates beauty filter, retouching, and animated effect APIs.
- Added retouching features including skin polishing, eye brightening, teeth whitening, wrinkle removal, and eye bag removal to the Enterprise Edition.
- Added the `onRemoteUserEnterRoom` and `onRemoteUserLeaveRoom` callbacks for the entry and exit of a user.

Optimization

- Optimized the PTS generation mechanism.
- Enabled automatic selection of the best access point after network change.
- Supported calling `startRemoteView` in advance.

Bug fixing

Fixed known crashes.

Version 6.7 Released on September 30, 2019

Added

- Added permission requesting configuration for AAR packaging.
- Supported CPU usage evaluation on Android 8.0 and above.

Optimization

- Sped up relayed push.
- Supported adjusting the playback volume of a specific user.

Version 6.6 Released on August 2, 2019

Added

- Supported local audio recording.
- Added callback APIs for sending the first audio and video frame.
- Added the system volume type setting API.
- Added the audio effect API for playing short audio effects.

- Made the data returned via the custom audio callback modifiable.

Optimization

- Sped up room entry and improved its success rate.
- Added an API to mute remote video.
- Unified room entry error codes, which are returned via `onEnterRoom` . If `result` is smaller than 0, it indicates failure to enter the room.
- Optimized the demo to support low-latency big rooms.
- Added a volume setting API and a volume callback API for the player.
- Supported local rendering for custom video publishing.
- Supported custom capturing and publishing of 1080p video.
- Supported the `SurfaceView` method for local and remote rendering.

Bug fixing

- Fixed the problems with relayed push and stream mixing.
- Fixed incorrect rotation for local preview.

Version 6.5 Released on June 12, 2019

Added

Added the "low-latency big room" feature for the live streaming mode (`TRTCApSceneLIVE`):

- Adopted a UDP protocol optimized for audio/video, allowing the SDK to better adapt to poor network conditions.
- Reduced the average watch latency to around 1 second, enhancing anchor-audience interaction experience.
- Supported rooms with up to 100,000 users.

Optimization

- Fixed lip-sync errors in poor network conditions.
- Optimized the `onStatistics` status callback. Callbacks are returned for only existing streams now.
- Optimized the playback buffer logic of `TXLivePlayer` , reducing stuttering.
- Sped up playback at the player end after local video is muted via `muteLocalVideo` and unmuted again.
- Optimized the QoE algorithm for high-latency and high-packet-loss network environments.
- Improved decoder performance and fixed increasing delay on earlier generations of Android phones.
- Optimized the volume evaluation algorithm (`enableAudioVolumeEvaluation`), improving accuracy.
- Optimized the QoE algorithm in the video call mode (`TRTCApSceneVideoCall`), further enhancing the smoothness of one-to-one calls in poor network conditions.

Bug fixing

- Fixed the occasional issue where no callback is returned for `enterRoom` .
- Fixed the issue where a user cannot play audio after disabling audio capturing.
- Fixed green screen after the local rendering view is removed and added again.
- Fixed the issue where the custom rendering callback (`setRemoteVideoRenderDelegate`) is returned only 10 times at most when the resolution of the remote video is 540p or above.

Version 6.4 Released on April 25, 2019

Added

- Added APIs for mirroring local video and encoded video.
- Added a callback for the `setMixTranscodingConfig` API.
- Added the eye enlarging, face slimming, chin slimming, and animated widget features to the Enterprise Edition.

Optimization

- Improved smoothness in poor network conditions.
- Optimized the volume callback algorithm, improving the accuracy of the values returned.
- Supported specifying data frame timestamps externally for the publishing of custom audio and video.
- Optimized the `setMixTranscodingConfig` API by adding the `roomId` parameter for stream mixing during cross-room co-anchoring.
- Optimized the `setMixTranscodingConfig` API by adding the `pureAudio` parameter for audio mixing and recording in audio-only call scenarios.
- Improved the 720p video decoding performance on low-end Android devices.

Bug fixing

- Fixed failure to switch to the hands-free mode.
- Fixed the occasional issue where live streaming (TXLivePlayer) latency increases and does not fall back.
- Fixed failure to use `setVideoEncoderRotation` in live streaming scenarios.
- Fixed the issue where no error callback is returned after the mic permission is denied on Android.
- Fixed the issue where a window pops up after the demo is opened on Android 9.0.
- Fixed failure by audience to adjust the volume using the volume buttons.

Version 6.3 Released on April 2, 2019

Added

- Supported 64-bit Android.
- Added a custom video capturing API: `TRTCCloud > sendCustomVideoData`.

- Added a custom audio capturing API: `TRTCCloud > sendCustomAudioData`.
- Added custom video rendering APIs: `TRTCCloud > setLocalVideoRenderDelegate + setRemoteVideoRenderDelegate`.
- Added a custom audio data callback API: `TRTCCloud > setAudioFrameDelegate`, which you can use to do the following:
 - Return the data captured by the mic: `TRTCAudioFrameDelegate > onCapturedAudioFrame`.
 - Return the audio data of each remote user: `TRTCAudioFrameDelegate > onPlayAudioFrame`.
 - Return the mixed audio data sent to the speaker for playback: `TRTCAudioFrameDelegate > onMixedPlayAudioFrame`.

Version 6.2 Released on March 8, 2019

Added

- Added the filter strength setting API `setFilterConcentration()`.
- Added the `sendSEIMsg()` API for sending custom messages through SEI headers in video frames. The feature is mainly used to insert timestamp information into video streams.
- Added the cross-room call feature `connectOtherRoom`, which allows two existing TRTC rooms to communicate with each other. This feature can be used to enable anchor competition across rooms.

Optimization

- Improved CPU utilization and stability.
- Enhanced video clarity in poor network conditions.
- Disabled the creation of multiple `TRTCCloud` instances and restricted instance creation to singletons. This can avoid cases where different instances of `TRTCCloud` compete for network resources, which compromise user experience.

Bug fixing

Fixed the problems with relayed push in audio-only call scenarios (such as Werewolf playing). You must specify the `bussInfo` field in `TRTCParam` to use the feature.

Version 6.1 Released on January 31, 2019

Optimization

- Supported watching screen sharing streams.
- Supported publishing custom video.
- Optimized CDN live streaming and stream mixing.
- Introduced two types of scenarios: live streaming and video calls, which are specified during room entry.
- Enhanced stability and fixed occasional crash.

- Optimized QoS control and improved performance in poor network conditions.

Version 6.0 Released on January 18, 2019

Optimization

- Updated the architecture to the LiteAV kernel.
- Adopted a new QoS algorithm, reducing stuttering and improving smoothness.
- Introduced a new audio module, enhancing audio quality in various network conditions.
- Supported dual-channel (primary stream and substream) encoding. We recommend you use this feature on Windows and macOS only.
- Supported CDN live streaming and stream mixing.

Release Notes (Web)

Last updated : 2022-09-23 18:06:13

A version number is in the format of `major.minor.patch`.

`major` : Major version number. If there is major version refactoring, this field will be incremented. Generally, the APIs of different major versions are not compatible with each other.

`minor` : Minor version number. The APIs of different minor versions are compatible with each other. If there is a new or optimized API, this field will be incremented.

`patch` : Patch number. If there is a feature improvement or bug fix, this field will be incremented.

notice

Please update to the latest version in a timely manner for service stability and better online support.

For notes on version updates, see [Update Guide](#).

Version 4.13.0 Released on July 8, 2022

New features

Added the [Client.destroy](#) API, complementing the lifecycle of [Client](#).

Improvements

Improved the encoding of the low-quality stream, delivering smoother playback.

Added an event for successful auto capturing resumption: [DEVICE_AUTO_RECOVERED](#).

Bug fixing

Fixed the issue where, after [Client.startMixTranscode](#) is called and the SDK is disconnected for longer than 30 seconds, stream mixing does not continue upon reconnection.

Fixed the issue where, after [Client.startPublishCDNStream](#) is called and the SDK is disconnected for longer than 30 seconds, publishing to CDN is discontinued even after reconnection.

Version 4.12.7 Released on June 17, 2022

Improvements

Added the `mixUser.renderMode` parameter to [Client.startMixTranscode](#). You can use it to set the rendering mode of the videos of users whose streams are mixed.

Changed the default value of `video profile` to `480p_2`, which uses less upstream bandwidth while delivering the same video quality. For details, see [LocalStream.setVideoProfile](#).

Improved the success rate of automatic playback resumption.

Improved the accuracy of the muting status after a disconnection and reconnection.

Fixed the issue where, on Safari 15.1, the page crashes when `muteVideo` is called. For details, see [webkit bug](#).

Bug fixing

Fixed the issue where the `stream-added` event is sometimes not thrown in the dual-stream mode (manual subscription).

Version 4.12.6 Released on June 10, 2022

Improvements

Added logic to prevent repeated room entry for `client.join`. For details, see [SDK Upgrade Guide](#).

Version 4.12.5 Released on May 20, 2022

Bug fixing

Fixed the issue where an error occurs when the npm package `trtc.umd.js` is loaded in Internet Explorer.

Fixed the occasional failure to play the low-quality stream after a subscription change.

Fixed the issue where, on Chrome 70 and earlier versions, moving the div container causes playback to pause.

Version 4.12.4 Released on May 7, 2022

Improvements

Sped up room entry.

Optimized the logic for switching to the low-quality stream.

Bug fixing

Fixed the occasional failure to invoke the `stream-added` event.

Fixed the failure to use Logitech cameras to capture 480p videos on Firefox.

Fixed the issue where, when the SDK is imported into WKWebView for iPad, an error occurs.

Version 4.12.3 Released on April 19, 2022

Improvements

Optimized the logic of high-resolution capturing on iOS 13 and 14.

Optimized the event listening logic to avoid cases where the SDK captures errors on the project side.

Added support for volume detection on Safari to help with dashboard troubleshooting.

Bug fixing

Fixed occasional failure to reconnect after a disconnection in the live streaming mode.

Fixed error getting the audio volume on iOS 11.

Version 4.12.2 Released on April 2, 2022

Improvements

Optimized the volume detecting logic, lowering memory usage and overhead.

Bug fixing

Fixed the issue where users are occasionally removed from the room (receive the `client-banned` callback) if they keep the SDK in the background for a long time.

Fixed the issue where, on iOS 15.2-15.4, echo is heard after camera switch. For details, see [WebRTC Known Issues and Solutions > Safari for iOS > Case 11](#).

Version 4.12.1 Released on March 18, 2022

Notes

See [Update Guide](#).

Improvements

You can now call the [stream.play](#) API multiple times and use it to turn on/off the mirror mode and set playback parameters in real time.

Optimized the auto capturing resumption logic to fix occasional failure to watermark published streams.

Bug fixing

Fixed the issue where, after the local user calls `muteVideo` or `unmuteVideo`, remote users see a black screen when trying to play the local user's small stream.

Fixed the issue where the `stream-subscribed` callback is received after users switch to the small stream.

Breaking change

Deprecated the `mirror` property of the [TRTC.createStream](#) API. Please use [stream.play\(elementId, { mirror: true }\)](#) instead.

Version 4.12.0 Released on March 4, 2022

Notes

See [Update Guide](#).

New features

Made [client.setRemoteVideoStreamType](#) an async API, which returns a promise that indicates whether the switch is successful.

Improvements

Improved scheduling accuracy for services outside the Chinese mainland.

Bug fixing

Fixed the occasional issue where a user is removed from the room due to `user_time_out` .

Version 4.11.13 Released on February 17, 2022

Improvements

Updated the TypeScript declaration file in the npm package.

Optimized the parameter authentication logic of [stream.play](#).

Bug fixing

Fixed the occasional issue where, on iOS 13, before access is granted, an error occurs when [LocalStream.initialize](#) is called.

Fixed the occasional issue where [AUDIO_VOLUME](#) returns 0.

Version 4.11.12 Released on January 11, 2022

Improvements

Published a TypeScript declaration file to the npm package.

Optimized the implementation logic of `stream.close()` .

Optimized the signaling logic for when `publish` / `unpublish` is called frequently.

Bug fixing

Fixed the issue where, on iOS 15.1, desktop webpages crash when streams are published. For details, see [WebRTC Known Issues and Solutions > Safari for iOS > Case 7](#).

Fixed the issue where [LocalStream.setAudioProfile\('high'\)](#) sets the bitrate to 192 Kbps.

Version 4.11.11 Released on December 17, 2021

Improvements

Optimized the auto capturing resumption logic, circumventing the issue of failure to resume capturing on some low-end Android devices.

Optimized the autoplay pop-up style.

Version 4.11.10 Released on December 03, 2021

Bug fixing

Fixed failure to disable the autoplay pop-up via `enableAutoPlayDialog: false`.

Fixed failure of the SDK to intercept repeated `stream.play` calls.

Version 4.11.9 Released on November 26, 2021

Notes

See [Update Guide](#).

Improvements

Supported displaying a pop-up window when autoplay fails. For details, see [Use the autoplay dialog provided by SDK](#).

Fixed the issue where **the SDK crashes whenever streams are published on iOS 15.1**. For details, see [WebRTC Known Issues and Solutions > Safari for iOS > Case 7](#).

To avoid the potential no audio issue, [TRTC.getMicrophones](#) no longer returns mics whose `deviceId` is `communications`. For details, see [WebRTC Known Issues and Solutions > Chrome > Case 8 & 9](#).

Optimized the `switchDevice` policy.

Improved the accuracy of the encoding/decoding support test in the context of WebView.

Improved parameter verification for [client.startPublishCDNStream](#), [client.stopPublishCDNStream](#), [client.startMixTranscode](#), and [client.stopMixTranscode](#).

Bug fixing

Fixed the occasional “TRTC not supported” error when `client.publish` is called.

Version 4.11.8 Released on November 5, 2021

Improvements

Circumvented the black screen issue during video playback on iOS 15.0. For details, see [WebRTC Known Issues and Solutions > Safari for iOS > Case 6](#).

Circumvented the issue where the SDK crashes whenever streams are published on iOS 15.1. For details, see [WebRTC Known Issues and Solutions > Safari for iOS > Case 7](#).

Version 4.11.7 Released on September 30, 2021

Improvements

Required parameter verification for key APIs.

Supported error messages in Chinese in the development mode (`LogLevel` set to [Debug](#)).

Improved the recovery success rate in cases of device capturing error.

Optimized the logic of call resumption after the system wakes up from hibernation.

Added `trtc.esm.js` and `trtc.umd.js` to meet the needs in different scenarios. For details, please see [TRTC Web SDK](#).

Version 4.11.6 Released on September 10, 2021

Improvements

Optimized the signaling scheduling logic, improving the success rate of room entry under poor network conditions. If you are using SDK v4.11.5, we recommend that you update to this version.

Version 4.11.5 Released on September 4, 2021

Improvements

Supported dynamic signaling channel scheduling, improving connection success rate under poor network conditions.

Supported cross-room stream mixing. For details, see [Client.startMixTranscode](#).

Bug fixing

Fixed occasional failure to receive the `stream-added` event callback after reconnection.

Fixed the occasional issue where the frame rate drops to 0 after screen sharing continues for a long time.

Version 4.11.4 Released on August 20, 2021

Improvements

Improved the accuracy of the H.264 support check for OPPO and vivo built-in browsers.

Supported auto capturing resumption (triggered in case of capturing error).

Added timeout logic for the `subscribe` API. For details, see the error code [API_CALL_TIMEOUT](#).

Bug fixing

Fixed occasional failure to pull streams on Safari for iOS in some older versions.

Fixed the issue where the mute status is inaccurate after the switching of devices.

Fixed the occasional issue where exceptions occur when the room entry API is called again after timeout.

Fixed the issue where, after a remote stream is unpublished, the audio/video player is not terminated in a timely manner.

Version 4.11.3 Released on July 30, 2021

Improvements

Optimized the exception handling logic for the `publish` and `subscribe` APIs.

Optimized the recovery policy for audio mixing plugins.

Bug fixing

Fixed the occasional inaccuracy of the `peer-leave` notification.

Version 4.11.2 Released on July 23, 2021

Improvements

Supported TURN server scheduling, improving connection success rate.

Added the `hasSmall` property, which indicates whether a remote user has substream video, to [Client.getRemoteMutedState](#).

Bug fixing

Fixed the issue where the SDK is unavailable when LocalStorage is disabled.

Fixed the occasional issue where API requests are not rejected when publishing exceptions occur.

Version 4.11.1 Released on June 25, 2021

Improvements

Supported the beauty filter plugin. For details, see [Enabling Beauty Filters](#).

Improved statistical accuracy.

Version 4.11.0 Released on June 18, 2021

New features

Supported the dual-stream mode. For detailed directions, see [Enabling Dual-Stream Mode](#).

Improvements

Optimized the event notification order.

Version 4.10.3 Released on June 11, 2021

Improvements

Optimized the quality measuring logic and allowed getting call quality statistics via a server-side API.

Added statistics on RTT and packet loss to [ClientEvent.NETWORK_QUALITY](#).

Optimized the API verification logic to prevent exceptions caused by repeated calls.

Optimized the playback logic, reducing audio loading time.

Version 4.10.2 Released on May 24, 2021

Improvements

Optimized the implementation logic of the `switchDevice` API, fixing occasional failure to switch to the front camera in Huawei Browser.

Increased the accuracy of the `StreamEvent.CONNECTION_STATE_CHANGED` event notification.

Bug fixing

Fixed occasional failure to play screen sharing streams on native applications.

Fixed occasional failure to receive the `stream-removed` event after reconnection.

Version 4.10.1 Released on April 30, 2021

New features

Added the `StreamEvent.CONNECTION_STATE_CHANGED` event for the change of stream connection status.

Added the `Client.getTransportStats` API, which can be used to obtain downstream RTT.

Supported using the `Client.getRemoteVideoStats` API to obtain statistics of the substream (screen sharing).

Improvements

Optimized the implementation logic of the `Client.switchRole` API.

Bug fixing

Fixed the issue where the `mute` event is occasionally triggered before `stream-added`.

Fixed the issue where no audio can be heard sometimes after room entry.

Version 4.10.0 Released on April 16, 2021

New features

Added the `client.startPublishCDNStream` API for publishing streams to the CDN of Tencent Cloud or a third-party vendor.

Added the `client.stopPublishCDNStream` API for stopping publishing streams to the CDN of Tencent Cloud or a third-party vendor.

Improvements

Optimized the parameter verification logic of the `localStream.switchDevice`, `localStream.addTrack`, and `localStream.removeTrack` APIs.

Version 4.9.0 Released on March 19, 2021

New features

Supported the preset layout mode for On-Cloud MixTranscoding. For details, see the [client.startMixTranscode](#) API.

Supported the callback of volume. For details, see the [client.enableAudioVolumeEvaluation](#) API.

Improvements

Changed the default port number of the WebSocket protocol to 443.

Bug fixing

Fixed the issue where audience cannot receive the callbacks of room entry and exit by an anchor in live streaming scenarios.

Fixed occasional failure to reconnect when string-type room IDs are used.

Breaking change

Supported returning detailed results of browser compatibility check via [TRTC.checkSystemRequirements](#). For details, please see the [API document](#) and [Update Guide](#).

Version 4.8.6 Released on March 1, 2021

Improvements

Supported stereo audio playback.

notice

Not supported on iOS yet.

Bug fixing

Fixed the issue where the `stream-removed` event is received when audio and video are disabled on a mobile device.

Version 4.8.5 Released on January 29, 2021

Improvements

Supported configuring multiple TURN servers via [client.setTurnServer](#).

Optimized the `userId` verification logic.

Bug fixing

Fixed the issue where the mute status is occasionally inaccurate after stream pushing starts.

Version 4.8.4 Released on January 15, 2021

Improvements

Supported dynamically calling the [localStream.setVideoProfile](#) API.

Optimized the data reporting logic of the dashboard.

Optimized the logic of dealing with autoplay restrictions. For details, please see [Suggested Solutions for Autoplay Restrictions](#).

Optimized the logic of dealing with failure to recover audio/video capturing after device connection/disconnection. For more information, please see [DEVICE_AUTO_RECOVER_FAILED](#).

Bug fixing

Fixed the issue where the mute status is occasionally inaccurate after stream pushing is resumed.

Version 4.8.3 Released on January 7, 2021

Improvements

Optimized the verification logic for the `roomId` parameter of the room entry API.

Bug fixing

Fixed the lack of third-party dependencies in version 4.8.2.

Fixed the issue where no audio is played when a user subscribes only to audio streams.

Fixed the occasional issue where no audio is played or `getAudioLevel` returns `0` after playback is resumed from autoplay restrictions on iOS.

Version 4.8.2 Released on December 31, 2020

Improvements

Optimized the verification logic for the `roomId` parameter of the room entry API. For more information, please see the [API document](#) and [Update Guide](#).

Optimized the timing of the `peer-join` and `peer-leave` notifications.

Bug fixing

Fixed the occasional `Cannot read property 'isConnected' of null` error during room exit.

Breaking change

Deleted the disused API `setDefaultMuteRemoteStreams`. Please use the `autoSubscribe` parameter of [TRTC.createClient](#) instead.

Version 4.8.1 Released on December 25, 2020

Bug fixing

Fixed occasional failure to hear remote users' audio on Windows.

Fixed the issue where the `client.getRemoteVideoStats()` API returns empty data.

Version 4.8.0 Released on December 18, 2020

New features

Supported On-Cloud MixTranscoding.

Supported string-type room IDs on all platforms. For details, please see the `useStringRoomId` parameter of [TRTC.createClient](#).

Allowed users to disable auto subscription. For details, please see the `autoSubscribe` parameter of [TRTC.createClient](#).

Improvements

Optimized the H.264 support check logic.

Optimized the device switching logic.

Optimized the status identification logic of the `hasAudio` and `hasVideo` APIs.

Bug fixing

Fixed occasional failure to reconnect after network disconnection.

Fixed black screen caused by frequent track adding or removing on Safari for iOS.

Version 4.7.1 Released on November 27, 2020

Improvements

Optimized the auto capturing resumption logic upon switch of media devices (which may be caused by a loose port or device plugging/unplugging).

Added the error code `DEVICE_AUTO_RECOVER_FAILED`, which indicates failure to restart a device.

Bug fixing

Fixed occasional black screen on Chrome 87.

Fixed the issue where, when both camera and screen sharing streams are pushed from a native application, users on web fail to pull screen sharing streams after subscribing to and unsubscribing from the streams repeatedly.

Version 4.7.0 Released on November 20, 2020

New features

Supported desktop Firefox M56+ and Edge M80+.

Improvements

Optimized the upstream bitrate control logic.

Optimized the retry logic for getting media devices.

Optimized the WebSocket reconnection logic.

Supported auto push resumption when a mic is plugged in/unplugged during audio mixing, optimizing the auto push resumption logic.

Breaking change

Supported returning detailed results of browser compatibility check via [TRTC.checkSystemRequirements](#). For more information, please see the [API document](#) and [Update Guide](#).

Version 4.6.7 Released on November 5, 2020

Bug fixing

Fixed the occasional issue of blurry screen during playback when hardware acceleration is enabled on Chrome.

Fixed failure to enter rooms and pull streams on WeChat's built-in browser for iOS.

Version 4.6.6 Released on October 23, 2020

Improvements

Optimized the retry logic for upstream peer connection.

Optimized the retry logic for downstream peer connection.

Optimized the logic of `TRTC.checkSystemRequirements`.

Supported screen sharing on Safari. For details, please see [Screen Sharing Guide](#).

Bug fixing

Fixed the issue where `getAudioLevel` returns `0` after audio playback is manually resumed due to autoplay restrictions.

Version 4.6.5 Released on October 14, 2020

Improvements

Optimized the logic for reestablishing WebSocket signaling channels, improving connection stability.

Optimized the log output logic.

Bug fixing

Fixed the issue where `getAudioLevel` returns `0` after resubscription on Chrome.

Fixed the issue where no audio is played after resubscription on Safari.

Fixed the issue where the `getLocalVideoStats` API returns `undefined` after the upstream audio track is replaced via `replaceTrack`.

Fixed occasional WebSocket disconnection upon change of network type during calls on a mobile device.

Version 4.6.4 Released on September 24, 2020

Improvements

Allowed stopping the collection of network quality statistics after room exit.

Bug fixing

Fixed the room entry error on Chrome 56.

Fixed the issue where image is rotated when relayed push is enabled on mobile devices.

Fixed on-cloud recording exceptions when audio-only streams are pushed.

Fixed failure to automatically resume push after the camera is unplugged due to inconsistent resolution.

Version 4.6.3 Released on August 28, 2020

Improvements

Optimized the compatibility check logic.

Optimized the log reporting logic.

Optimized the upstream bitrate control logic.

Version 4.6.2 Released on August 14, 2020

Improvements

Optimized the upstream bitrate control logic.

Optimized the `switchRole` parameter verification logic.

Optimized the logic of measuring upstream network quality.

Optimized error messages.

Supported automatically resuming push when change of the stream capturing device is detected.

Bugs fixed

Fixed failure to publish again immediately after `unpublish` succeeds.

Version 4.6.1 Released on July 28, 2020

Improvements

Supported checking via `TRTC.isScreenShareSupported` whether screen sharing is supported. Safari does not support screen sharing.

Optimized the parameter verification logic for the `subscribe` and `unsubscribe` APIs.

Added network quality logs.

Bugs fixed

Fixed the “OverconstrainedError” error when access to media devices is not granted and an empty device ID is passed in the `TRTC.createStream` API.

Fixed the issue where no log is printed when upstream peer connection is lost.

Version 4.6.0 Released on July 16, 2020

New features

Added the `NETWORK_QUALITY` event.

Version 4.5.0 Released on July 2, 2020

New features

Added the `screenAudio` parameter to the `createStream` API.

Bugs fixed

Fixed the issue where echo cancellation does not work in browsers for Android.

Fixed the issue where the RTT value returned by the `getTransportStats` API is `NAN`.

Version 4.4.0 Released on May 28, 2020

New features

Supported capturing system audio (on Windows) or audio of the current tab (on macOS) during screen sharing on Chrome 74 or above.

Version 4.3.14 Released on April 29, 2020

Bugs fixed

Fixed the `muted` and `unmute` events for Mini Program.

Version 4.3.13 Released on April 16, 2020

Improvements

Optimized the availability check logic.

Version 4.3.12 Released on April 13, 2020

Bugs fixed

Fixed a potential `RTCPeerConnection` status change exception.

Version 4.3.11 Released on March 28, 2020

Improvements

Supported detection of mobile QQ Browser, which does not support TRTC SDK for desktop browsers at the moment.

Bugs fixed

Fixed Boolean return types.

Version 4.3.10 Released on March 17, 2020

Improvements

Optimized the environment detection logic.

Added `name` `code` for `RtcError` .

Version 4.3.9 Released on March 13, 2020

Improvements

Supported automatic detection of the deployment environment.

Optimized logging.

Version 4.3.8 Released on February 24, 2020

Improvements

Added the `streamId` and `userdefinerecordid` parameters to `createClient` .

Version 4.3.7 Released on February 21, 2020

Improvements

Fixed the issue where an exception is throw upon device switch during screen sharing.

Bugs fixed

Fixed the device occupation issue by releasing MediaStream after device switch.

Fixed the potential error of the subscription API.

Version 4.3.6 Released on February 25, 2020

Bugs fixed

Adjusted the audio/video playback sequence of `Stream.resume()`, and fixed autoplay exceptions in WeChat's built-in browser on iOS.

Version 4.3.5 Released on February 5, 2020

Improvements

Added timeout check for the `publish` API, improving the success rate of signaling.

Version 4.3.4 Released on January 6, 2020

Improvements

Updated `core-js` to version 3.6.1.

Bugs fixed

Fixed the issue where an exception is thrown after `unpublish` times out.

Fixed the issue where third-party libraries cause decline in the performance of V8.

Version 4.3.3 Released on December 25, 2019

Improvements

Added the ability to check whether the environment supports WebRTC.

Optimized the SDP response mechanism.

Optimized the reporting logic.

Bugs fixed

Optimized the TURN URL protocol format.

Version 4.3.2 Released on December 9, 2019

Improvements

Supported automatic reconnection after downstream ICE disconnection.

Removed the STUN hole punching step to increase the success rate and speed of user connection via private networks.

Used the server-calibrated UTC time as the timestamp for log reporting.

Optimized ICE error reporting.

Added more key events to `avmonitor` .

Bugs fixed

Fixed the 1005 reconnection error of WebSocket signaling channels.

Fixed the issue with the reporting of downstream packet loss rate.

Version 4.3.1 Released on November 23, 2019

Improvements

Supported automatic reconnection when upstream ICE disconnects during calls.

Bugs fixed

Fixed the issue where the host ICE candidate of public IP type does not take effect after STUN hole punching fails.

Version 4.3.0 Released on November 15, 2019

New features

Added the `client.getTransportStats()` API.

Improvements

Made log reporting more detailed.

Supported wildcard characters for event unbinding.

Extended the connection timeout threshold to 5 seconds.

Extended the publishing timeout threshold to 5 seconds.

Bugs fixed

Fixed the issue of abnormal SDK judgment due to modification of the prototype chain of `zone.js` .

Version 4.2.0 Released on November 4, 2019

New features

Added the `client.off()` API, which can be used to unbind client events.

Improvements

Optimized the collection of call status statistics.

Added permission check for `client.publish()` .

Added an auto playback error prompt for `stream.play()` and `stream.resume()` .

Bugs fixed

Fixed black screen when the camera is switched via `localStream.switchDevice()` .

Version 4.1.1 Released on October 24, 2019

Bugs fixed

Fixed log loss.

Fixed the loss of remote users who reconnect after disconnection.

Version 4.1.0 Released on October 17, 2019

New features

Supported passing object `HTMLDivElement` in the `stream.play()` API.

Supported setting audio attributes via `localStream.setAudioProfile()` . Currently, two profiles are supported: standard and high.

Bugs fixed

Fixed the restriction on the number of WebAudio Context on Chrome.

Fixed the issue where the local audio/video player is not restarted after `replaceTrack()` .

Fixed the issue where custom settings via `localStream.setScreenProfile()` do not take effect.

Fixed the issues with the restart of the audio/video player and status reporting.

Version 4.0.0 Released on October 11, 2019

Provided APIs in the Client/Stream format, allowing for clearer role assignment and naming.

The new version is not compatible with the legacy version. In addition to APIs, the new version introduced the following changes:

Changed the method of setting video attributes. All video attributes (resolution, frame rate, and bitrate) are now set using the `localStream.setVideoProfile()` API of the SDK via applications. The new version does not support setting video attributes via “Image Settings” (Spear Role) in the Tencent Cloud console.

Integrated an audio/video player into the stream object. Playback is now solely controlled by the SDK.

Supported subscribing to and unsubscribing from remote audio/video streams via the `client.subscribe()` and `client.unsubscribe()` APIs.

Release Notes (Electron)

Last updated : 2022-09-23 18:07:23

Version 9.3.201 Released on January 5, 2022

New features

Windows & macOS: added the [onSpeedTestResult](#) callback, which returns the result of network speed testing.

Improvements

Windows & macOS: improved the performance of the speed testing API [startSpeedTest](#).

Windows & macOS: added the `streamType` parameter to the [muteLocalVideo](#) API (which is used to pause/resume publishing local video).

Windows & macOS: added the `streamType` parameter to the [muteRemoteVideoStream](#) API (which is used to pause/resume receiving a remote video stream).

Windows & macOS: added the `source`, `captureRect`, and `property` parameters to the [selectScreenCaptureTarget](#) API (which is used to configure screen sharing).

Windows & macOS: added the `params` parameter to the [startScreenCapture](#) API (which is used to start screen sharing).

Bug fixing

macOS: fixed the camera video capturing issue on macOS 12.

Windows & macOS: optimized the QoS control policy under poor network conditions to enable smoother communication.

Windows: improved the AGC algorithm, reducing cases of excessively low or high volume.

Windows: fixed the frame rate exception for the capturing of screen sharing images.

Version 8.9.102 Released on August 11, 2021

New features

Windows & macOS: added the new parameter `gatewayRtt` to the [onStatistics](#) callback.

Bug fixing

macOS: fixed crash caused by logging on special devices.

macOS: fixed the issue where, after `setAudioCaptureVolume(0)` is used to mute audio, the mic testing volume is 0.

Windows: improved performance and fixed the issue of black screen after the camera is turned on.

Windows: fixed the issue where the resolution is not restored after being automatically reduced during screen sharing.

Windows & macOS: fixed other bugs.

Version 8.6.101 Released on May 28, 2021

New features

Windows & macOS: added APIs for excluding windows from screen sharing: [addExcludedShareWindow](#), [removeExcludedShareWindow](#), [removeAllExcludedShareWindow](#).

Windows & macOS: added the `isMinimizeWindow` field to the return code of the sharable window enumerating API [getScreenCaptureSources](#).

Windows & macOS: supported passing constructor functions as parameters to APIs.

Bug fixing

Windows: fixed the issue where paths containing Chinese characters are not supported for plugin loading.

Windows & macOS: fixed the `webgl context lost` issue.

Windows & macOS: fixed the issue where the images of remote users freeze after the local user switches to the small image (dual-channel encoding enabled).

Windows & macOS: fixed the issue where, when a user enters the room and starts pulling streams, the images of remote users are blurred before they gradually become clear.

v8.4.1 Released on March 26, 2021

New features

macOS: supported capturing system audio via [startSystemAudioLoopback](#), i.e., the system loopback feature that is enabled on Windows. The feature allows the SDK to capture system audio so that anchors can stream local audio or video files to other users.

macOS: supported callback of the system audio capturing result via [onSystemAudioLoopbackError](#), which allows you to learn about the status of the system audio driver.

macOS: supported local preview for screen sharing. You can now display screen sharing preview in a small window.

All platforms: supported the beauty filter plugin architecture.

Quality improvement

All platforms: improved audio quality in the [music](#) mode, which makes it more suitable for Clubhouse-like audio streaming scenarios.

All platforms: improved the adaptability to poor network conditions. Smooth audio and video can be delivered even when the packet loss rate reaches 70%.

Windows: improved audio quality in some streaming scenarios by significantly reducing audio damage.

Windows: improved performance by 20%-30% in some scenarios.

Bug fixing

macOS: fixed the issue where, after the screen sharing user switches to desktop sharing and then back to the sharing of a specific window on Mac mini (M1), remote users still see the user's desktop.

macOS: fixed the issue where the shared content is not highlighted (on macOS 11.1 and 10.14.5, there isn't a green border around the shared content; on macOS 10.3.2, the green border is displayed, but the shared window flickers when maximized).

macOS: fixed the issue where, when users on Mac mini (M1) get the screen sharing list, the SDK crashes because `sourceName` is null and "" is returned.

macOS: fixed the issue on Mac mini (M1) where, when `getCurrentMicDevice` is called, the SDK crashes because `sourceName` is empty.

Windows: fixed the issue where the SDK crashes when the desktop is shared on Windows Server 2019 Datacenter x64.

Windows: fixed the issue where screen sharing sometimes ends unexpectedly when the target window is resized during screen sharing.

Windows: fixed image capturing failure with some cameras.

v8.2.7 Released on January 6, 2021

New features

Windows & macOS: added [switchRoom](#) to switch rooms.

Windows & macOS: added [setLocalRenderParams](#) to set rendering parameters for the local image (primary stream).

Windows & macOS: added [setRemoteRenderParams](#) to set rendering parameters for a remote image.

Windows & macOS: added [startPlayMusic](#) to play background music.

Windows & macOS: added [stopPlayMusic](#) to stop background music.

Windows & macOS: added [pausePlayMusic](#) to pause background music.

Windows & macOS: added [resumePlayMusic](#) to resume background music.

Windows & macOS: added [getMusicDurationInMS](#) to get the total length of the background music file, in milliseconds.

Windows & macOS: added [seekMusicToPosInTime](#) to set the playback progress of background music.

Windows & macOS: added [setAllMusicVolume](#) to set the audio mixing volume of background music.

Windows & macOS: added [setMusicPlayoutVolume](#) to set the local playback volume of background music.

Windows & macOS: added [setMusicPublishVolume](#) to set the remote playback volume of background music.

Windows & macOS: added the [onSwitchRoom](#) callback for room switching.

Windows & macOS: added [setRemoteAudioVolume](#) to set the playback volume of a remote user.

Windows & macOS: added [snapshotVideo](#) to take a video screenshot.

Windows & macOS: added the [onSnapshotComplete](#) callback for the completion of a screenshot.

Improvements

Windows & macOS: supported string-type `strRoomId` for the `enterRoom` and `switchRoom` APIs.

Windows & macOS: fixed other bugs.

v7.9.348 Released on November 12, 2020

Improvements

Windows: supported the use of paths containing Chinese characters to save recording files.

Windows: supported the anti-covering feature in the window capturing area.

v7.8.342 Released on October 10, 2020

New features

Windows & macOS: added the [onAudioDeviceCaptureVolumeChanged](#) callback for volume change of the current audio capturing device.

Windows & macOS: added the [onAudioDevicePlayoutVolumeChanged](#) callback for volume change of the current audio playback device.

v7.7.330 Released on September 11, 2020

New features

Windows & macOS: added [setAudioQuality](#) to adjust audio quality.

Improvements

Windows: fixed the issue of high CPU utilization when some low-end cameras are used.

Windows: optimized the compatibility with multiple USB cameras and mics to make it easier to turn on such devices.

Windows: optimized the selection policy of cameras and mics to avoid audio/video capturing exceptions caused by the connection/disconnection of cameras and mics.

Windows & macOS: fixed other bugs.

v7.6.300 Released on August 26, 2020

New features

Windows & macOS: added [setCurrentMicDeviceMute](#), [getCurrentMicDeviceMute](#), [setCurrentSpeakerDeviceMute](#), and [getCurrentSpeakerDeviceMute](#) to control mics and speakers on PC.

v7.5.210 Released on August 11, 2020

Improvements

Windows & macOS: fixed the issue where SDK callbacks are not in sequence.

Windows & macOS: fixed the issue where switching rendering modes causes crashes.

Windows & macOS: fixed the issue where rendering fails for certain resolutions.

Windows & macOS: fixed other bugs.

v7.4.204 Released on July 01, 2020

Improvements

Windows: optimized the acoustic echo cancellation (AEC) effect on Windows.

Windows: improved the compatibility with cameras on Windows.

Windows: improved the compatibility with audio devices (mics and speakers) on Windows.

Windows: fixed the issue where the `UserID` returned by `onPlayAudioFrame` is incorrect on Windows.

Windows: supported system audio mixing on 64-bit Windows.

v7.2.174 Released on April 20, 2020

Improvements

macOS: fixed occasional resolution inconsistency for local custom rendering on macOS.

Windows: optimized the `getCurrentCameraDevice` logic on Windows to return the first device as the default device when the camera is not used.

Windows: fixed the issue where the highlighted window is displayed as a gray screen during screen sharing.

Windows: fixed the issue where the system occasionally freezes when users get screen share thumbnails on Windows 10.

Windows & macOS: fixed the issue where the custom stream ID occasionally fails to take effect immediately after role switching.

Windows & macOS: fixed the issue where the encoding parameters of screen sharing do not take effect.

Windows: fixed the issue where it takes a long time for a screen shared by a Windows user to be displayed to a WebRTC user.

v7.1.157 Released on April 02, 2020

New features

Supported [screen sharing](#) via the [primary stream](#).

Improvements

Improved the usability of [preset stream mixing templates](#).

Increased the success rate of [stream mixing](#).

Optimized screen sharing on Windows.

v7.0.149 Released on March 19, 2020

New features

Added the [trtc.d.ts](#) file for TypeScript developers.