IntPowers

TECHNICAL DOCUMENT · CONNECTIVITY · CCaaS

Network connectivity requirements

Widgets, video kiosks, apps and agents on your CCaaS platform

Video RTC
WebRTC video calls

Your CCaaS
SPLIT integration over SIP

Customer: **[CUSTOMER]**

What each endpoint needs

There are two types of connection: signaling and media. Users only need RTC and ICE; agents also need their CCaaS network.

RTC

Video call signaling and media through the video gateway.

ICE

STUN/TURN servers to traverse NAT and firewalls.

CCaaS

Agent voice, routing and desktop, as required by your CCaaS vendor.

ENDPOINT	WHERE IT IS USED	RTC	ICE	CCaaS
Web widgets	Click-to-Call and Click-to-Video on the customer website	Yes	Yes	—
Video kiosks	Touch or QR kiosks in branches, hotels and stores	Yes	Yes	—
Mobile apps	Applications built with the Mobile SDK	Yes	Yes	—
CCaaS agents	CCaaS desktop with the embedded video panel	Yes	Yes	Yes

Checklist

Five steps to get the customer network ready before go-live.

1. **Allow the services**
   Open the RTC and ICE services on the network of each endpoint and, for agents, the CCaaS network.
2. **Check the edge, if applicable**
   With the Global Low Latency edge, confirm the applicable destinations with IntPowers.
3. **Ensure bandwidth**
   Meet the minimum bandwidth and the network quality thresholds.
4. **Apply best practices**
   TLS and SRTP encryption, RTP ranges and recommendations for agents.
5. **Validate the connection**
   Test from the same computer and network with Video RTC Tester.

Service details

Customer: **[CUSTOMER]**

CCaaS: **[PLATFORM]**

Network destinations: see slide 8

Tester: **[TESTER URL]**

Email: support@ivrpowers.com

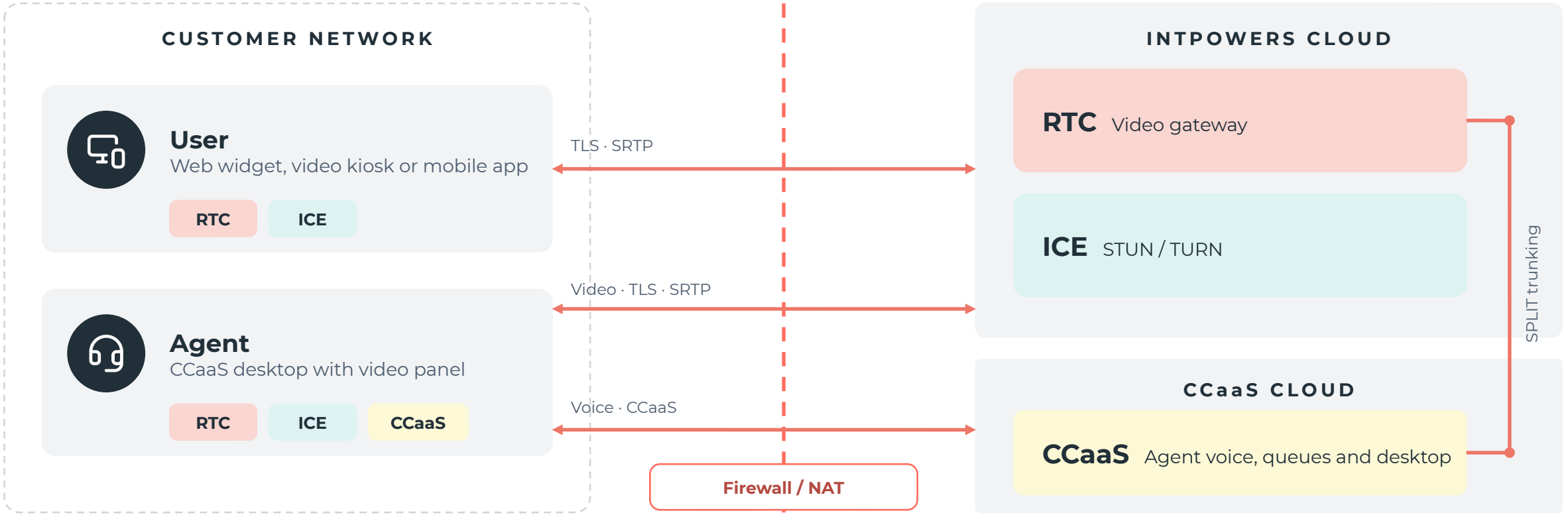
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How each endpoint connects

Signaling is encrypted with TLS and audio and video media with SRTP.



The gateway delivers each video call to the CCaaS as a SIP call: its queues route it and the agent answers from their desktop. Video arrives through the embedded panel (RTC and ICE).

CCaaS agents: RTC, ICE and CCaaS network

The video panel needs RTC and ICE to the destinations on slide 8; voice and desktop need your CCaaS network.

DESTINATION	PROTOCOL	PORTS	USE
RTC	TCP	`80 · 443`	HTTP (redirected to HTTPS) · HTTPS
	TCP	`8989`	WebSocket to the video gateway (recommended)
	TCP	`8089`	HTTPS-API to the video gateway (alternative)
	UDP	`19099–20099`	RTP media for the video panel [1]
ICE	TCP · UDP	`80 · 443`	STUN/TURN (443 over TLS)
	UDP	`19099–20099`	RTP media [1]
CCaaS	—	`Per vendor`	Agent desktop, signaling and voice [2]
			Genesys Cloud: see the appendix on slide 10

[1] Video panel media can be relayed through STUN/TURN to traverse NAT, but this is not recommended for agents. [2] IntPowers does not manage the CCaaS network: apply the current requirements published by your vendor.

Widgets, video kiosks and apps: RTC and ICE

Users only need RTC and ICE. On home or mobile networks, no configuration is usually required.

DESTINATION	PROTOCOL	PORTS	USE
RTC	TCP	80 · 443	HTTP (redirected to HTTPS) · HTTPS
	TCP	8989	WebSocket to the video gateway (recommended)
	TCP	8089	HTTPS-API to the video gateway (alternative)
	UDP	19099–20099	RTP audio and video media [1]
ICE	TCP · UDP	80 · 443	STUN/TURN (443 over TLS)
	UDP	19099–20099	RTP media [1]

Video kiosks on corporate, hotel or shopping mall networks: validate these ports with the site network manager before installation.

[1] With a restrictive firewall, the RTP range can be skipped: media is then relayed through STUN/TURN on ports 80 and 443.

Bandwidth and network quality

Values per participant and per direction. The codec adapts to the available network at all times.

Minimum bandwidth

MODE	UPLOAD	DOWNLOAD	CODEC
Audio only	40 kbps	40 kbps	Opus
Audio and video	80 kbps	80 kbps	Opus · VP8

Quality thresholds

PARAMETER	MAXIMUM
Round-trip latency (RTT)	< 300 ms
Delay variation (jitter)	< 50 ms
Packet loss	< 1.5%

Recommended for good video quality

- 512 kbps upload and download per endpoint. Minimum values allow the call, not HD quality.
- Agents: wired connection and no other heavy downloads during the shift. CCaaS voice usage comes on top of these values.

Destinations to allow and best practices

IntPowers provides the destinations for each service. They are filled in for each customer before this document is sent.

DESTINATION	DOMAIN OR IP ADDRESS	PORTS
RTC	**[RTC DESTINATION]**	Slides 5 and 6
ICE (STUN/TURN)	**[ICE DESTINATION]**	Slides 5 and 6
CCaaS	Per vendor (Genesys Cloud: slide 10)	Slide 5
Global Low Latency edge	**[EDGE DESTINATION · IF APPLICABLE]**	Per service

If the service does not use the Global Low Latency edge, remove that row.

Best practices

- Signaling encrypted with TLS and media encrypted with SRTP.
- Allow the RTP range on agent workstations: relaying their media through STUN/TURN is not recommended.
- With a restrictive firewall, users can work through STUN/TURN on 80 and 443 only.
- Apply the network requirements published by the CCaaS vendor on agent workstations.

Check before you start

Always test from the same computer and network that will be used in production.

1

Test with Video RTC Tester

Open **[TESTER URL]** in the workstation browser and check that every test completes.

2

Agents: test a full session

Place a test video call to the service queue and answer it from the CCaaS desktop.

3

If anything fails, send the report

Send support the tester results, the time of the test and the interaction ID in the CCaaS.

End-to-end test

- Video RTC Tester completes on the workstation.
- The video call enters the CCaaS queue.
- The agent answers from their desktop.
- Two-way audio, through the CCaaS.
- Stable video in the embedded panel.

Acceptance criteria per agent workstation

Projects on Genesys Cloud

Genesys WebRTC phone on the agent workstation, in addition to RTC and ICE (slide 5). Remove this slide if the CCaaS is a different one.

DESTINATION	PROTOCOL	PORTS	USE
Genesys Cloud	TCP	443	HTTPS: desktop and WebRTC phone
	UDP	3478	STUN: required (TCP 3478, reserved)
	UDP	16384–65535	SRTP and TURN audio media
	UDP	19302	Google STUN (optional)

Genesys publishes the IP addresses and domains for each region in "IP addresses for the firewall allowlist" and "Domains for the firewall allowlist".

Source: Genesys Cloud Resource Center, "Ports and services for WebRTC phones", checked October 2026. The current version published by Genesys always prevails.

SPLIT trunking and video panel

- The video gateway delivers each session to your Genesys organization as a SIP call (SPLIT).
- IntPowers or your partner configure SPLIT trunking. With the gateway in the IntPowers cloud, the customer network is not involved.
- With an on-premises gateway, allow SPLIT trunking SIP signaling to the Genesys Cloud public IPs for your region.
- The video panel is embedded in the Genesys desktop and uses RTC and ICE.

SUPPORT AND DOCUMENTATION

Need help?

Technical support

Email: support@ivrpowers.com
Help Center: support.ivrpowers.com
Phone ES: +34 911 599 846
Phone US: +1 786 329 7177

CCaaS interoperability

How Video RTC integrates with your platform, with a dedicated Genesys module.

Vendor documentation

Network requirements for the agent desktop and voice, published by your CCaaS.