Web Services and Applications Configuration Guide

WebRTC

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WebRTC

> **Important**
> login.voice.prompt-dn-less-phone-number is mandatory for WebRTC.

Workspace Web Edition provides the following options for integrating WebRTC:

web-rtc.enable-dtmf-tone

- Default Value: `true`
- Valid Values: `true, false`
- Changes take effect: When the session is started or restarted.
- Description: When this option is true, a tone is played when the agent selects a DTMF key.

web-rtc.gateway-uri

- Default Value:
- Valid Values: A valid URI
- Changes take effect: When the session is started or restarted.
- Description: Defines the host for the WebRTC gateway.

web-rtc.identifier.x-last-digit-displayed

- Default Value: `10`
- Valid Values: A positive integer
- Changes take effect: When the session is started or restarted.
- Description: Specifies the number of digits of the WebRTC identifier to display in Workspace. The WebRTC identifier is generated based on the user ID — for example, a user ID of a78b5fc6bc7742fb9802958c985098e3 is converted to 977898510299698997742102989802958999850981013. You can use this option to make the number a more user-friendly size when it's displayed in the interaction for the toolbar, parties in Case Information, and in Team Communicator. For example, if you leave the option at the default value, the WebRTC identifier above would be displayed as 9850981013 in Workspace.

web-rtc.phone-number-prefix

- Default Value:
- Valid Values: Any string
- Changes take effect: When the session is started or restarted.

- Description: Specifies the prefix that is used for the phone number that is dynamically created for WebRTC.

web-rtc.quality-alert-timeout

- Default Value: `30`
- Valid Values: A positive integer
- Changes take effect: When the session is started or restarted.
- Description: Time in seconds that passes before an alert is displayed to notify that timeout occurred during a test call.

web-rtc.quality.score.key-name

- Default Value: `callQualityScore`
- Valid Values: Any string
- Changes take effect: When the session is started or restarted.
- Description: Specifies the key in the UserEvent that Workspace should use for the call quality score set by the agent. The agent only provides this score if the WebRTC API detects low quality of audio for the call.

web-rtc.quality.statistics.key-name

- Default Value: `callQualityStatistics`
- Valid Values: Any string
- Changes take effect: When the session is started or restarted.
- Description: Specifies the key in the UserEvent that Workspace should use for the call quality statistics it receives from the WebRTC API.

web-rtc.request-quality-interval

- Default Value: 2
- Valid Values: A positive integer
- Changes take effect: When the session is started or restarted.
- Description: Specifies the interval, in seconds, to request audio call quality from the Genesys WebRTC JavaScript API.

web-rtc.ringing-bell

- Default Value:
- Valid Values: All special characters that are valid URL file path, '|' separator and numeric values.
- Changes take effect: When the session is started or restarted.
- Description: Specifies the voice channel ringing sound-configuration string. For example: 'BELL|7|0'. The value has three components that are separated by the pipe ('|') character. The first is the sound file

name (the BELL, RING, CHORD or WARNING predefined aliases or an absolute URL to a MP3 file). The second is a priority — the higher the integer, the higher the priority. The last is the duration, which can have the following values:

- `-1` — play and repeat the sound until an explicit message, such as event established, causes it to stop.
- `0` — play the whole sound once
- an integer greater than `0` — the length of time, in milliseconds, to play and repeat the sound

web-rtc.stun-uri

- Default Value:
- Valid Values: A valid URI
- Changes take effect: When the session is started or restarted.
- Description: The URI to the STUN server.

web-rtc.troubleshooting.enable-feedback

- Default Value: `true`
- Valid Values: `true, false`
- Changes take effect: When the session is started or restarted.
- Description: Enables WebRTC to send a user event that contains diagnostic information.

web-rtc.turn-password

- Default Value:
- Valid Values: A valid password
- Changes take effect: When the session is started or restarted.
- Description: The password for the TURN server.

web-rtc.turn-username

- Default Value:
- Valid Values: A valid username
- Changes take effect: When the session is started or restarted.
- Description: The username for TURN server.

web-rtc.turn-uri

- Default Value:
- Valid Values: A valid URI
- Changes take effect: When the session is started or restarted.

- Description: The URI to the TURN server.

privilege.web-rtc.can-mute-microphone

- Default Value: `false`
- Valid Values: `true`, `false`
- Changes take effect: When the session is started or restarted.
- Description: Allows an agent to mute and unmute the microphone. Depends on privilege.voice.can-use and privilege.web-rtc.can-use.

privilege.web-rtc.can-mute-speaker

- Default Value: `false`
- Valid Values: `true`, `false`
- Changes take effect: When the session is started or restarted.
- Description: Allows an agent to mute and unmute the speaker. Depends on privilege.voice.can-use and privilege.web-rtc.can-use.

privilege.web-rtc.can-send-dtmf

- Default Value: `true`
- Valid Values: `true`, `false`
- Changes take effect: When the session is started or restarted.
- Description: Specifies if DTMF is available for WebRTC calls. Depends on privilege.voice.can-use and privilege.web-rtc.can-use.

privilege.web-rtc.can-use

- Default Value: `false`
- Valid Values: `true`, `false`
- Changes take effect: When the session is started or restarted.
- Description: Enables WebRTC in Workspace. Depends on privilege.voice.can-use.