AudioCodes WebRTC Solution for Contact Centers

WebRTC

The **AudioCodes WebRTC** solution is a highly secure, comprehensive, and easy to deploy web calling solution that delivers improved customer experience, cost savings and enhanced customer service operations.

AudioCodes' WebRTC solution comprises:

- a feature-rich, highly secure WebRTC gateway fully integrated with AudioCodes market-leading session border controller (SBC)
- a WebRTC client SDK, enabling quick and easy integration of WebRTC into client websites and smartphones.

The AudioCodes WebRTC solution provides enterprise IT administrators and contact center operators with a highly secure, comprehensive, easy to deploy, web-based voice communications solution that results in improved customer experience, cost savings and enhanced customer service operations.

Enhanced IT productivity and smoother customer service operations

Savings in communications and infrastructure costs

Simple and secure integration of WebRTC into enterprise and contact center operations

Simple integration of WebRTC functionality into websites without prior coding knowledge

Features

- ✓ Fully integrated with AudioCodes SBCs for robust security and interoperability
- ✓ Seamless connectivity between WebRTC clients and VoIP deployments
- ✓ Click-to-call widget for simple addition of click-to-call buttons on websites
- ✓ High quality voice with built-in Opus support
- ✓ Support for real-time voice quality monitoring

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WebRTC Gateway

Platform	VMware	KVM	AWS	Mediant 500/800	Mediant 4000	Mediant 9030	Mediant 9080
Number of WebRTC sessions	2,700	3,500	3,000	100	1,000	5,000	8,000
Number of WebRTC Opus to G.711 transcoding sessions	325	3,500	3,000	54 (Mediant 800C)	1,000	1,050 integrated or 5,000 with transcoding clusters	1,050 integrated or 8,000 with transcoding clusters

Fully featured, single instance WebRTC gateway. The WebRTC gateway is integrated with the SBC and includes both signaling and media capabilities.

- ICE Lite support
- DTLS and SRTP support
- SIP over WebSocket
- WebRTC recording (using standard SIPREC protocol)
- High availability

- Audio codec transcoding
- Signaling transport – WebSocket/UDP/TCP/TLS, IPv4/IPv6
- SIP interoperability
- Call detailed records and performance monitoring

- RTP <-> SRTP transrating
- OAuth, SIP digest user authentication
- SIP registrar
- Voice quality monitoring

Client WebRTC SDK for Web, iOS and Android

Supported mobile platforms	Web (HTML5), iOS and Android
Supported features	Voice and video calls, outgoing and incoming calls, chat
Client authentication	SIP Digest and OAuth

Web SDK

Supported browsers	Chrome, Firefox, Safari, Edge
Session rehydration	Hitting refresh page (F5) on browser causes only a short pause in a call, after which the call is resumed.
Logging	Callback API for detailed SDK logs
Geo-redundancy	SDK hops between SBCs if one is not available
	SIP over WebSocket/SIP Register
	DTMF/Hold/Retrieve/Transfer/Call waiting/Delayed-offer/3Pcc

Web click-to-call widget. SDK wrapper allows click-to-call button integration into a website with just a few lines of code.

iOS and Android SDK

Native SDK (not a 'Web View' implementation) for optimal user experience

Push notification for incoming calls