Webcall Overview

- High reliability and performance with carrier-grade availability. The fastest, most scalable and most dependable.
- Interoperable between any WebRTC device and any SIP network using SIPEX10 a MIRK new Technology.
- Comprehensive, secure authentication, encryption, and attack mitigation. Centralized Servers Monitoring.
- Powerful and scalable signalling platform and media engine in dedicated secure data and voice servers.
- SDK for rapid real time communication application development.

Challenges

● while the WebRTC protocol is designed to enable simple peer-to-peer web communications, for WebRTC to become a widely used communications technology, sophisticated server-side applications must be developed to address:

• Application control and synchronization during network changes and browser page reloads • Rapid application integration with existing systems • Identity management between multiple devices and across web and telephony domains • Border and application security to prevent attacks and service abuse • High capacity media handling for NAT traversal, encryption, and transcoding • Robust and dynamic interworking with existing secured infrastructure.

Solution Overview

Webcall UK company with a DEV LAB in Belgrade Serbia has created the Webcall Communications WebRTC Session Controller to address these challenges and provide a foundation for innovation. The Oracle Communications WebRTC Session Controller is a highly available, carrier grade solution designed to enable rapid development and deployment of powerful and differentiating WebRTC applications. Communications with WebRTC Session Controller is based on proven technologies from Webcall Communications service delivery platform product. It brings carrier grade network capabilities into the web domain enabling CSPs and enterprises to create a new form of high quality and privacy communications for their users and staff.

Service	Benefit
Reliability, Rehydration, Handover One to One.	Automatically re-establishes dropped Webcall WebRTC sessions, including dropped network signal, browser reset, network handover or user-initiated swap of devices with extended SIPEX10
Scalability, Availability, High Security level.	Distributed high-availability signalling and media architecture for carrier-grade scalability Highly scalable, software-based SIPEX10 termination for privacy and high security level.
Browser and Application Interoperability	Rapid development with extensible JavaScript Client SDK provides automatic browser mediation, client authentication, SIPEX10 sessions and connection self-management in our sophisticated platform.
Network and Encryption Interoperability	Bridge Webcall with WebRTC to existing networks with WebRTC to SIP/IMS signalling and WebRTC media to existing Voice over IP system. Network based encryption, codec transcoding and multi-vendor interoperability
Users and Networks Security	Choice of user authentication mechanism: Web-based user authentication (OAuth) or traditional Telco/enterprise authentication At the network level, prevents both overload of the edge and back-end infrastructure as well as DoS attacks while prioritizing traffic to maintain normal service to valid users. Efficiently handles encryption keys & network authentication

Summary

Using WebRTC, CSPs may create new web-based communication services and extend existing services to web-based clients. Enterprises can extend access to their UC and contact center communications infrastructures to mobile users. The distinctive advantages of the Webcall Communications Web Session Controller provides reliability, interoperability, and security. The Webcall Communications WebRTC Session Controller reliably maintains an active session through browser anomalies or network failures. It incorporates web-based security standards providing network security, authentication, and authorization. With a focus on interoperability, the Webcall Communications WebRTC Session Controller provides signalling, IP address and media interworking to support large scale, reliable interoperable, universal communications.

As the leading global Webcall provider WebRTC-based SIPEX10 Session Controller Mirk Technology enables Communications Service Providers (CSPs) and enterprises to offer Webcall services – from any device, across any World Wide Network – with carrier-grade reliability, privacy and security.

Webcall Voice Communications are increasingly shifting to web and IP-based applications that are outside of traditional Public telephony networks. Enterprise users want to access their unified communications applications with their own Internet-connected mobile devices and consumers increasingly prefer Internet-based communications channels to access contact centers.

Contact Us For more information about the Webcall Global Communications using WebRTC-based Technology Session Controller, visit allwebcall.com or Click to Call us button to speak to a representative.

WEBCALL GLOBAL COMMUNICATIONS WebRTC-based SIPEX10 EXTENDED SESSION CONTROLLER