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Delivering Enterprise-Class Communications with WebRTC

Executive Overview

WebRTC is an emerging industry standard for enabling Web browsers with real-time communications capabilities. It enables enterprises to enhance Web sites, empower BYOD users, and improve video collaboration and on-line meetings, to name but a few examples. An open standard, WebRTC helps IT organizations accelerate time to market and contain costs by breaking vendor and platform dependencies. And it enables better user experiences by eliminating cumbersome browser-specific vendor application plug-ins.

WebRTC introduces a variety of connectivity, security, and control challenges for corporate IT planners. New tools are required to bridge the Web world with the enterprise VoIP and unified communications (UC) world and to ensure secure and reliable enterprise-class communications. This paper explores potential WebRTC applications, reviews WebRTC deployment considerations, and explains how WebRTC service enablers – a new type of network element – help enterprises reap all the benefits of browser-based communications without compromising security, reliability, or service quality.

Real time communications over the Web is a cornerstone in Oracle's portfolio of solutions for the hyper-connected enterprise. Offering rich infrastructure solutions for WebRTC and SIP-based communications, Oracle enables enterprises to become hyper-connected, where users are always on-line and always collaborating – from home, office, and the road.

WebRTC Brings Real-Time Communications to the Browser

WebRTC (Web Real-Time Communications) is an open standard for embedding real-time multimedia communications capabilities directly into a Web browser. Prior to WebRTC, there were no standard methods for enabling interactive communications in a browser environment. Developers relied on plug-ins like Adobe® Flash® or custom browser extensions that required cumbersome downloads. Poor user experiences combined with a lack of standards and multivendor interoperability have inhibited the widespread adoption of browser-based communications.

WebRTC is a standard drafted by the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF) to overcome these adoption barriers. The open standard framework eliminates the need for special-purpose client software and onerous plug-ins and downloads. Instead, interactive voice, video, and data-sharing functions are delivered as standard components of the Web developer's toolkit. Ordinary Web developers, who aren't necessarily versed in telephony, can create multimedia communications-enabled applications using simple HTML and JavaScript APIs. End-users enjoy an improved experience with no interruptions for downloads, consistent operation across devices and browsers, and immersive communications capabilities.

WebRTC is expected to foster a broad ecosystem of interoperable applications and services. Leading browser providers such as Google (Chrome™), Mozilla® (Firefox®) and Opera™ include WebRTC engines in their latest releases. And many communications solution providers and ISVs are leveraging WebRTC in enterprise, contact center, and customer service applications.

Browser-Based Communications Eliminates Cost and Complexity

By breaking vendor and platform dependencies, WebRTC has the potential to fundamentally transform enterprise communications. Until now, businesses have been held hostage to expensive PBX desk phones and proprietary softphone clients. While legacy IP-PBX and UC vendors support open standards such as SIP, many lock in customers and maximize product margins by reserving full-feature support for proprietary endpoints and separately licensed softphone clients.

With more and more workers using smartphones as their primary handset, expensive PBX desk phones are becoming increasingly difficult to justify. What businesses need instead is a way to make smartphones and tablets full-fledged alternatives to traditional PBX phones. But existing solutions for extending enterprise communications services to mobile devices are costly and inefficient. Most UC vendors offer operating-system-specific soft clients that take time and money to qualify, deploy, and support.

WebRTC overcomes these limitations by bringing real-time communications directly to the browser, eliminating special purpose, OS-specific clients. With WebRTC, IT organizations can accelerate time-to-market and contain costs by efficiently extending enterprise communications services to any browser-enabled device – smartphone, tablet, or PC. Users can access the WebRTC-enabled service over any network – public or private; WiFi, mobile broadband, or wired LAN.

WebRTC reduces upfront IT expenses by containing client licensing fees, qualification efforts, and deployment costs. There are no proprietary clients to purchase, roll out, update, or support. The client application runs on an off-the-shelf "free" browser. Qualification, deployment, and maintenance costs are contained to the Web site. New features and fixes are implemented right on the Web page.

Real-Time Communications Improves Customer Exchanges and Mobile Productivity

WebRTC enables a new generation of platform-independent, lightweight business communications applications that boost employee productivity and improve customer interactions. Sample WebRTC-enabled business applications include:

- **Customer-facing Web sites** – add voice, video, and screen-sharing to storefronts, support sites, or mobile apps. Convey contextual information to contact center applications to streamline workflows and improve customer interactions. Give agents visibility into Web page data and prevent customers from repeating information.

- **Mobile VoIP and UC clients** – extend corporate communications services and productivity tools to tablets and smartphones for Bring Your Own Device (BYOD) or fixed mobile convergence (FMC) initiatives. Eliminate proprietary UC client expenses and support burdens by bringing real-time communications directly to the browser. Reduce international mobile roaming and calling fees with Wi-Fi calling. Move sessions seamlessly between Wi-Fi and mobile broadband for FMC.

- **Communications-enabled business processes** – add real-time communications to Web-based business applications. Streamline workflows, eliminate human latency, and improve decision making by weaving interactive voice, video, or real-time data channels into business processes.

- **On-line meeting applications** – introduce lightweight video collaboration and screen-sharing tools for on-line meetings, Webinars, and customer presentations. Eliminate bulky client plug-ins that delay meetings, waste time, and impede collaboration.

WebRTC Enterprise Deployment Considerations

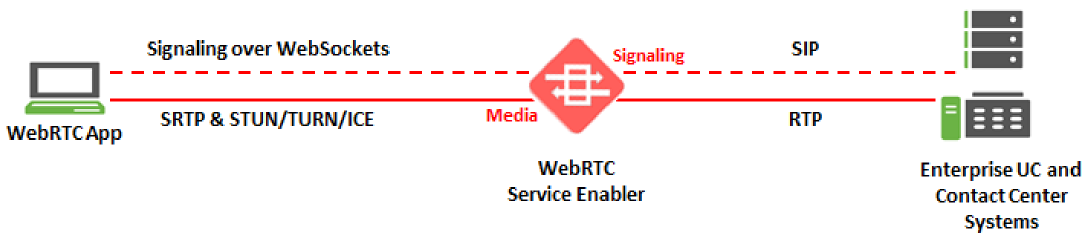
Enterprises are likely to introduce WebRTC applications in an incremental manner. Most will start by adding browser-based applications to an incumbent UC or IP-PBX environment. There are a variety of interoperability challenges and functional issues to consider when tying WebRTC features into the existing enterprise UC or VoIP infrastructure.

First and foremost, WebRTC is defined for simple peer-to-peer communications between browsers. It is not a framework for implementing a full-fledged enterprise unified communications system. The WebRTC specifications focus primarily on the media framework. The signaling plane – call setup and session management functions – is left to the application developer. The WebRTC specifications do not address authentication, authorization and accounting (AAA) functions, or specify any standard mechanisms for leveraging corporate directory services or policy stores such as Microsoft® Active

Directory®. Nor do they include provisions for ensuring reliable communications – mitigating network problems or protecting against denial of service attacks.

New server-side network elements are required to bridge the Web and corporate UC worlds and to ensure enterprise-class reliability for browser-based communications. Some vendors are promoting WebRTC gateways to tie browser-based applications into enterprise UC environments. But most gateways only provide basic stateless protocol translation functions. For example they may convert SRTP media streams to RTP media streams, or convert JavaScript constructs to SIP signaling flows. Gateways do not address the inherent scalability, reliability, or security issues associated with bringing WebRTC applications to the enterprise.

A new product category has been conceived to tackle these challenges. A *WebRTC service enabler* integrates the Web and corporate UC worlds, while enabling enterprise-grade service quality.



A WebRTC service enabler provides seamless interoperability with enterprise-grade reliability and security

WebRTC service enablers are specifically designed to extend enterprise VoIP and UC services to the browser. Unlike a simple WebRTC gateway, a WebRTC service enabler provides:

- Strong security to protect signaling and media flows associated with WebRTC endpoints
- Stateful session management (rehydration[1]) to maintain session continuity in the event of network timeouts and hand-offs, browser resets, or page reloads
- Comprehensive session control to protect against service overloads and to ensure high service quality

[1] See http://tools.ietf.org/search/draft-ietf-rtcweb-jsep-02

- Media anchoring to enable fixed mobile convergence, move calls between endpoints, facilitate integration with enterprise UC elements (e.g. session recorders, IVRs, media servers), and other advanced service orchestration functions
- Flexible protocol interoperability to interwork diverse signaling and media methods
- Support for enterprise AAA systems, directory services, and policy stores
- NAT and firewall traversal capabilities
- Fully redundant implementations to ensure continuous service availability in the event of equipment, interface, or network failures.

WebRTC service enablers let enterprises enjoy all the benefits of browser-based communications – platform and network independence, faster service deployment, lower costs – without sacrificing security, availability, or service quality.

Conclusion

WebRTC has the potential to dramatically improve enterprise communications. By eliminating platform and vendor dependencies, and bringing real-time voice, video, and collaboration directly to the browser, WebRTC can enrich customer experiences and increase employee productivity while eliminating cost and complexity. But the WebRTC framework addresses only simple peer-to-peer communications. New server-side solutions are needed to interwork the Web and enterprise VoIP and UC domains. WebRTC service enablers were specifically conceived to extend corporate VoIP and UC services to the browser. WebRTC service enablers let businesses unleash the full potential of browser-based communications without compromising security or reliability.

About Oracle Communications WebRTC Session Controller

The Oracle Communications WebRTC Session Controller is a carrier-class WebRTC service enabler that lets businesses extend UC and contact center communications to any user with a Web browser. It provides dynamic media anchoring, supports standards-based identity management including Oauth, and offers comprehensive session rehydration functionality, delivering seamless Web-to-SIP network interoperability and enterprise-grade reliability and security.

Based on field-proven technologies from the Oracle family of enterprise communications solutions, the Oracle Communications WebRTC Session Controller enables rapid deployment of innovative

WebRTC applications that span devices and networks. The comprehensive solution includes robust signaling and media engines for interworking the Web and SIP domains and an extensible SDK for accelerating client development.

Oracle enterprise communications solutions are enabling the hyper-connected enterprise with a communications architecture that seamlessly connects fixed and mobile users to each other, enabling rich multimedia customer interactions and automating business processes for significant increases in productivity, efficiency, and ROI.

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