Hewlett Packard Enterprise

Business white paper

Innovating with WebRTC

WebRTC Applications and Enablement

WebRTC enables native integration of IP communications services into web browsers, without an additional plug-in. It's flexible and does not mandate any signaling protocol for call control or communications, providing a high degree of flexibility for integration in IT environments. This works for real-time voice and video sessions and messaging communications services.

Create value with WebRTC

Review WebRTC

The Web real-time communications (WebRTC) standard—an emerging HTML5 technology—is a major innovation in the way communications services and applications are delivered. Associated with other disruptions—such as LTE, VoLTE, or network functions virtualization (NFV)—it promises to significantly enhance the user experience and also deeply impact communications service provider (CSP) business models. Indeed, by combining the native compliance of a large base of enabled devices with the wide reach of the web developer community—along with a complete set of communications tools, WebRTC enables rich communications to become a standard web capability, accessible far beyond the closed world of telecoms.

WebRTC standardizes how real-time communications—voice, video, data, text, or a blend of these—are accessed and delivered through a simple web browser across any device. By integrating communications services as part of the web browsing experience, WebRTC removes one of the most important barriers to seamless multimedia communications and unleashes new types of services and uses. No longer is there a need to download a plug-in or separate application for mobile Internet protocol (IP) communications. Entering into a voice or video conversation is at the fingertip of any website—blending video, voice, and more advanced capabilities, such as document sharing, becomes truly intuitive.

Beyond the amazing user experience it enables, WebRTC also significantly changes the perspective for developers and communications service providers. It's based on JavaScript, standard APIs, and standard codecs, making it easy for the web developer community and traditional telecom community to grasp. This, in itself, extends the reach of telecom enablers into the enterprise application development ecosystem.

The size of the telecom developer community worldwide comprises a few tens of thousands of programmers, while the JavaScript® developer community extends to several millions. Bringing embedded telecom services to the fingertips of this ecosystem will foster faster innovation and accelerate its adoption into the fabric of more traditional enterprise applications. As a result, by securely opening up their communications resources to a larger ecosystem and giving access to a wider range of communication devices, CSPs can monetize assets more extensively.

For CSPs, WebRTC is a unique opportunity to re-establish themselves as the central player in the value chain for delivering communications services to users—a position that's been threatened by Over-the-Top (OTT) providers in recent times. With WebRTC, CSPs can:

- Directly enhance customer offerings. For example, enable customers to use a wide range of devices for the same account by creating a much richer user interface, or by extending reach beyond cellular networks, leveraging, for example, Wi-Fi networks.
- Much more easily embed rich communications capabilities into other CSP services, such as vertical applications, call centers, and prepay offerings.
- Create a secure and rich environment for third-party developers, enabling asset optimization while monetizing new aspects of the networks, such as quality of service (QoS), presence, preferences, and storage.
- Make it possible to provision optimized and virtual core network instances, dedicated to specific use cases for mobile virtual network enablers (MVNEs), OTT players, or large corporations, for example. In this case, WebRTC provides the right framework to open and dedicate specific network instances in a secure manner and with the appropriate service-level agreements (SLAs) to institutional entities.

The WebRTC Application and Enablement solution

The WebRTC Applications and Enablement solution, that includes the HPE WebRTC Gateway Controller, integrates Real Time Communication in any web page, such as chatting with an expert and switching to Audio/Video calls without need for any plug-ins. Examples include click-to-call, multi-device communication services, screen sharing, video sharing, call recording, audio and video conferencing, voice mail access, and unified communication.

WebRTC opens new possibilities to address vertical applications and use cases, due to:

- Its capabilities to make user experiences more intuitive
- The simplicity of integrating the technology into standard web environments

Make remote inspection and diagnostics
In a world increasingly dominated by complexity, there's often a need for expert support—at work or home, on the go, and in all kinds of situations. But the physical availability of such an expert might be problematic, costly, or slow, especially with the development of e-commerce and other types of virtual or remote commercial transactions.

For example, a customer buys a device online and notices defects on delivery or finds it hard to boot up without remote assistance. Or a small car accident occurs and the insurance company requires some visual of the damage. At a factory, a field technician might require the expertise from a specialized engineer who understands a particular problem happening on a sensitive piece of equipment.

Traditionally, such cases have been handled in inefficient or costly ways. Examples include:

- Sending an expert to the place where the issue occurred.
- Replacing parts that could be fixed remotely.
- Having angry customers, who after spending hours on the phone with remote agents, are still hampered by limited information and left unsatisfied.

As a result of such experiences, customer satisfaction is often low, and the cost can be high.

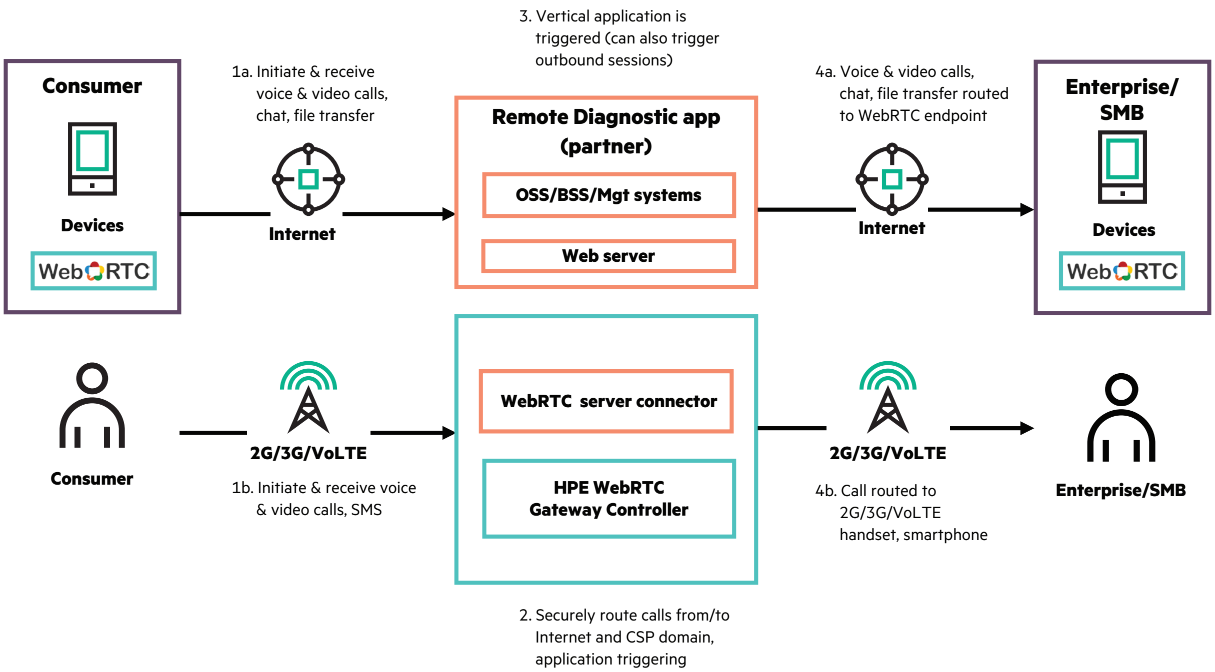

Figure 1. WebRTC remote diagnostic

WebRTC technology enhances these processes by creating a seamless and multimodal communication channel between the expert and the user. A remote diagnostics application—customer specific or third party, with the **HPE WebRTC Gateway Controller**, is designed to provide a feature-rich environment to leverage the power of WebRTC for such situations.

Gain a multimodal virtual contact center

The multimodality brought by the generalization of web technologies and ubiquitous connectivity—and the evolution of how we use the web fostered by the Web 2.0 revolution—all contributed to profoundly change the way enterprises manage their customer relationships. In the past, customer contact happened mostly over the phone or through an interactive voice response (IVR), with touch points being an unusual event. Now, a much richer customer experience is expected; today clients and vendors expect a 24/7 relationship, with faster-than-light interactivity and content-rich contexts.

WebRTC can enhance the customer experience and significantly improve customer relationship management by providing advanced contact center capabilities.

For example, by providing an easy click-to-connect interface, HPE WebRTC Gateway Controller can implement an efficient front end to various contacts within the enterprise using flexible rules. This implements a more efficient alternative to traditional, directly connected automatic call distribution systems or IVR front ends.

To illustrate the added value that WebRTC brings, consider the usual situation where an IVR fronts the enterprise's main contact center, prompts the caller to choose the reason for the call—commercial request or technical support, and then redirects and parks the call until an agent becomes available. In some jurisdictions, such as in the European Union, call waiting time can no longer legally be charged to the calling party. This creates additional costs and resource use issues for the enterprise. WebRTC can eliminate this problem, as it supports establishing calls only at the time when both parties are available to take them. Figure 2 shows an example of a call flow that implements this functionality.

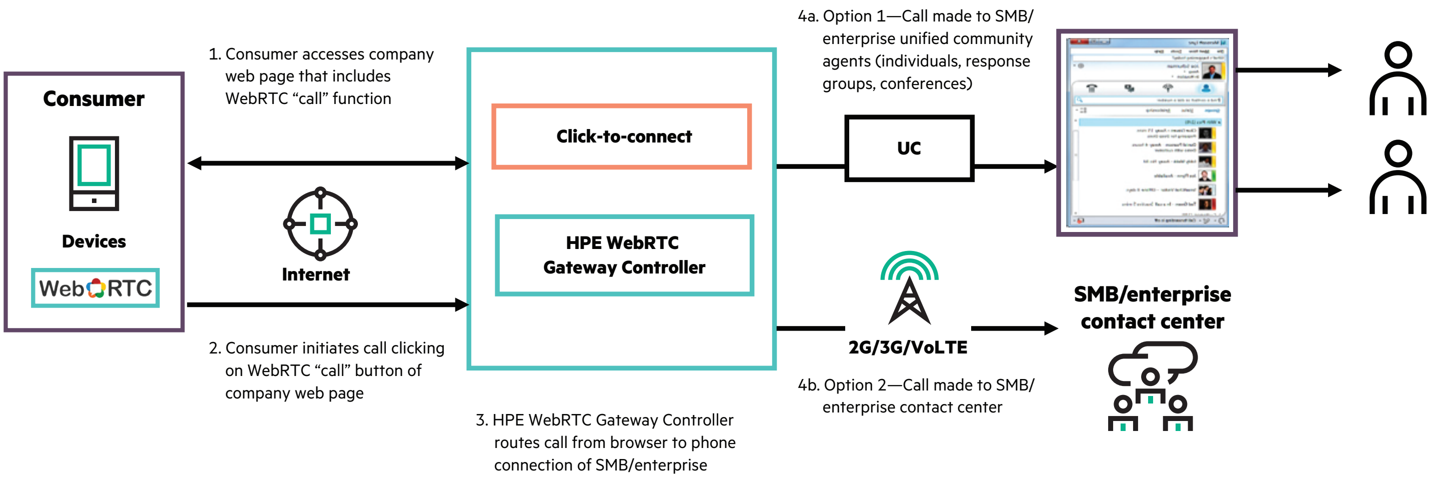

Figure 2. WebRTC click-to-connect to customer care scenario

WebRTC can significantly improve the transfer of contexts to contact centers in a seamless fashion. For example, a quick response or QR code can be used to encode a WebRTC URL, which is used to directly call the customer contact center in charge of supporting a given product, together with the product's unique identifier. The unique QR code can be generated dynamically at purchase time. So, if the customer experiences a problem with a product, they can scan the QR code, which will automatically generate a WebRTC call to the contact center, transferring the product information, together with additional information like language. This is done without installing or configuring any additional software.

Once contact is established, WebRTC also enables a rich customer experience across a variety of modes and devices. When using a contact or call center, a major challenge stands out: How to manage incoming requests from chat and a hotline at the same time? Today, systems are designed in silos and lead to increased costs and resources—human, technological, and equipment.

Sponsored calls

The idea of sponsored calls—monetizing calls through advertisements pushed to the caller, rather than charging users directly—has been around for a long time. It was developed further with the emergence of OTT players, whose business models heavily rely on advertising. But so far, technical constraints and poor ergonomics have limited the success of such services.

A CSP willing to offer sponsored calls to customers using WebRTC will typically:

- Give access to a web portal—normally the CSP's web portal
- Get customers authenticated, and let them place free calls—from a WebRTC portal

In return, the customer accepts the ad insertions within the WebRTC services. These ads can be personalized with multiple parameters or insight, such as device used, location, social profile, and credit status.

In addition, several value-added features can be provided as part of these sponsored calls, such as:

- The subscriber's calling number is not presented to the called party or is replaced by a virtual number that has a limited validity period or through other ways to represent the caller's identity.
- The unique token generated to make sure the user watched the ad can provide extra information, such as credit—number of free minutes, available audio, video quality, and more.

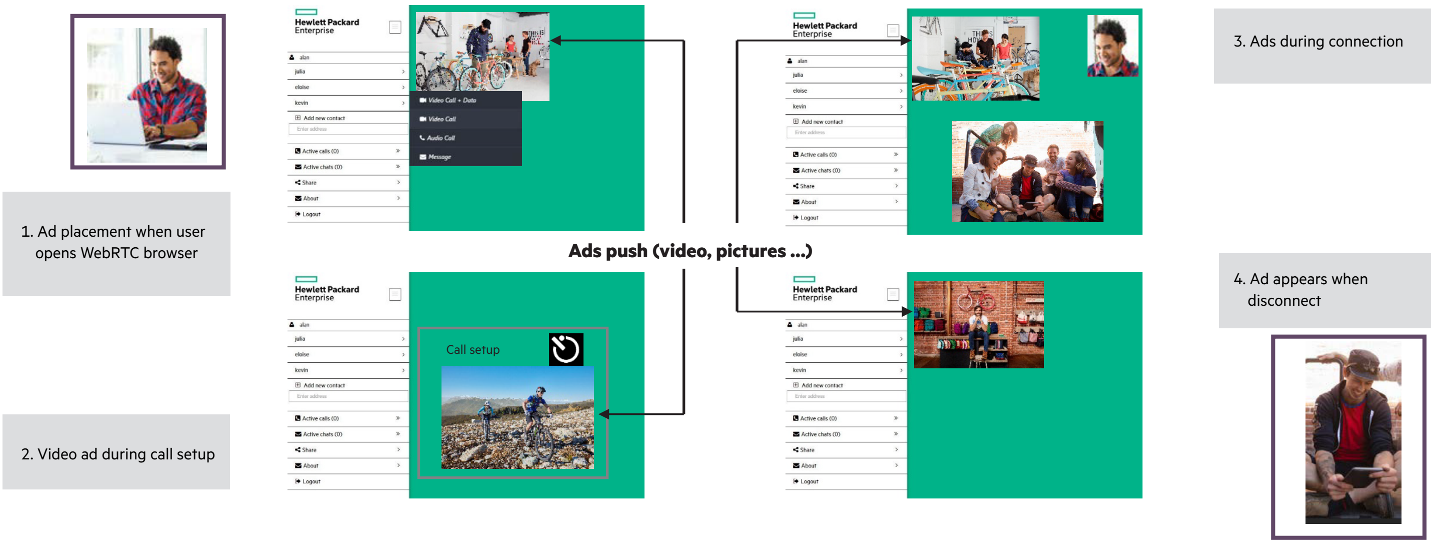

Figure 3. Sponsored-calls experience

WebRTC provides a new and promising way for CSPs and enterprises to address these market opportunities. Indeed, by providing easy options for mixing various media streams coming from different sources, WebRTC removes most of the technical issues that were attached to the relatively inflexible environments of the past. It also provides much more appealing ways to display advertising messages, which can even enhance the user's experience. For example, short video clips can be pushed to the caller while the call is being set up, filling in what would otherwise be idle waiting time.

This type of scenario enables CSPs to address specific markets, such as the traditional prepay, and improve the total customer experience, creating more stickiness with their customer bases—improving top-line and retention ratios.

Learn about WebRTC APIs and development platform for CSPs

WebRTC makes it possible to simply and securely establish and manage voice and video calls. CSPs can use this capability to create new applications on their own, or enable third-party applications. In this latter case, the WebRTC exposure might be integrated into a more comprehensive application development framework, which would enable important functionality such as access control, service governance, real-time policy management, customized billing, SLA management, or application-development tools.

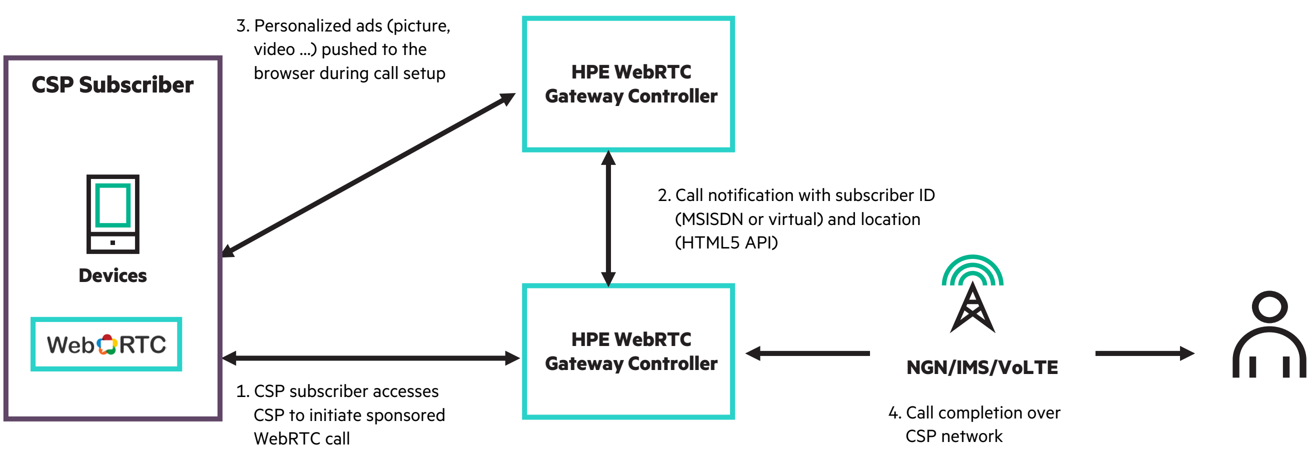

Figure 4. High-level view of sponsored-calls use case

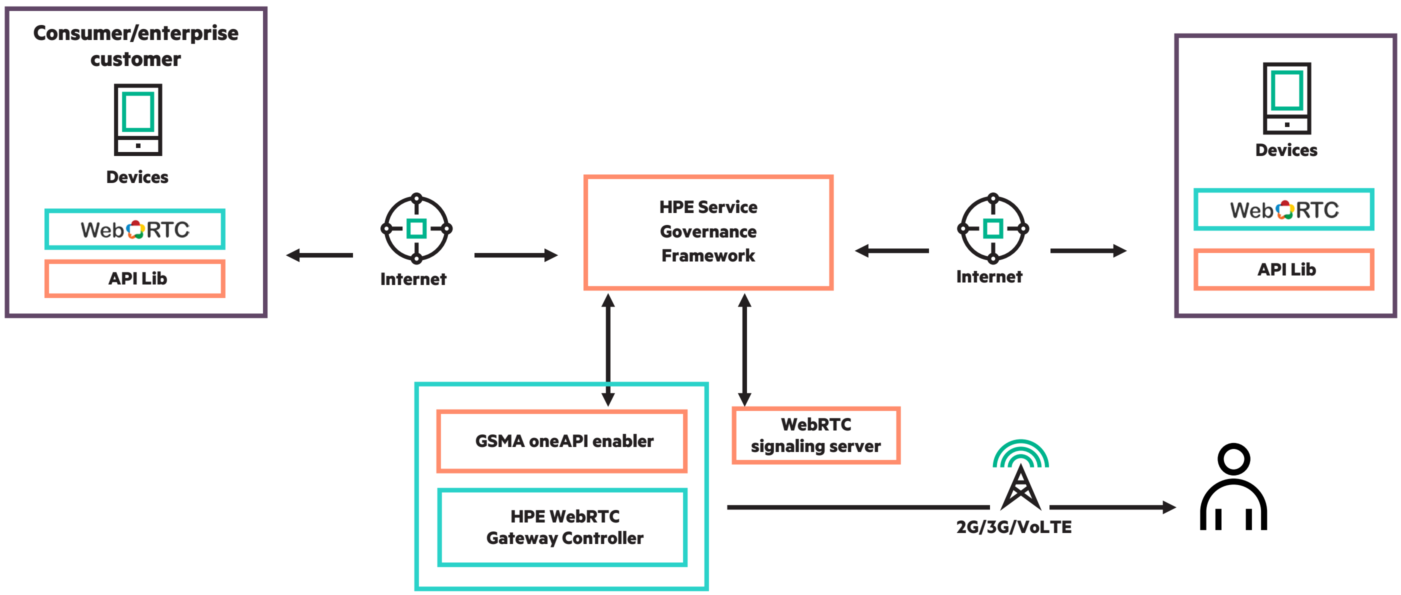

Figure 5. WebRTC API and development platform overview

With such a complete framework, CSPs can give access to their network resources, while developing a profitable ecosystem of applications and partners and keeping full control of their assets. For example, CSPs can monetize WebRTC use differently, depending on the application profile. Also, they can monitor and guarantee SLAs, and seamlessly leverage other components, such as customer profiles or Big Data repositories, or develop a vivid ecosystem of developers.

Figure 5 highlights the various components required to provide a WebRTC development platform.

In operations, a typical call flow would follow this pattern:

- The customer initiates a multimedia session by clicking on the WebRTC “call” button within a third-party application.
- The WebRTC connection request is sent to the HPE API Management interface, which performs a predefined procedure:
  - Check the incoming requestor’s profile and determine a number of parameters that influence the context of the execution. For example, the initiator might have a premium subscription, enabling high-definition video calls to go through; or on the contrary, the initiator might have a subscription that only allows voice calls.
  - Organize and execute the chaining of subsequent actions, which depend on the context retrieved during the previous step and application business logic. For example, customers with a valid premium subscription will be connected to a video conference service with high-definition quality and high priority in the network. They will also be able to use the document-sharing option.
  - Use the API management function to take care of potentially complex charging and settlement activities.

As described here, WebRTC can be complemented with sophisticated features, enabling developers to fully leverage capabilities from the network and implement contextual business logics.

Know the benefits

WebRTC is more than technology. It promises to change the way people and applications communicate by:

- Enabling contextual communications within the fabric of any personal or business application
- Removing barriers among communications modes—such as voice, text, video, devices, and networks
- Opening up communications networks to a much wider community of developers and a new set of innovative applications
- Enabling new business models for CSPs

This Hewlett Packard Enterprise market-leading solution gives CSPs a robust, efficient, scalable, and secure set of capabilities to address the WebRTC opportunity.

Underpinned by HPE leadership in IT services, infrastructure, and cloud, this solution makes it possible to:

- Create opportunities for new revenue immediately—and with improved customer stickiness—with a set of off-the-shelf WebRTC innovative applications from Hewlett Packard Enterprise and its partners
- Build an ecosystem of application developers and other partners to expand the reach beyond traditional markets
- Reduce implementation and security risks—due to a robust, proven, and scalable platform, based on industry standards

Get Voice over Wi-Fi with WebRTC

As LTE networks get deployed, mobile network operators (MNOs) are looking at ways to improve the coverage and quality of their networks, while improving the efficiency of their network operations. Voice over Wi-Fi (VoWi-Fi) is a popular way to achieve these objectives. While roaming on a Wi-Fi network, customers can use high-bandwidth Wi-Fi access points instead of LTE networks, resulting in cheaper costs for the user and operator.

This also provides a low-cost alternative to international roaming, and may guarantee better coverage indoors. By offering a better continuity of service across access methods, and improving the overall quality of service due to better network offload, VoWi-Fi also enables mobile network operators to reduce their churn rate and gain a competitive advantage over OTT players.

For mobile virtual network operators (MVNOs), VoWi-Fi can generate significant business value. For example, it reduces dependence on specific mobile network operators, decreasing costs and providing light-weight, market-entry strategies.

Most of the time, users can access VoWi-Fi services on their communications devices through an application in the Apple® App Store® or Google Play™ store. This works well, but creates some limitations and barriers to adoption, as the application has to be downloaded and installed on the device to make calls; and it has to run in the background to receive calls.

Now, WebRTC provides a new and more universal alternative. It only requires a web browser be installed on the device, which is a default option. By removing the need for potentially cumbersome manipulation of local apps and settings, WebRTC makes the VoWi-Fi option readily available at any time and from any device—significantly improving its adoption by default.

Learn more at
hpe.com/csp/netapps

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