WebRTC IMS SYSTEMS AND WebRTC PROPRIETARY ISLANDS

INNOVATIONS ENABLEMENT OR PARADIGM SHIFT?

TECHNICAL WHITE PAPER

WebRTC is a game-changing technology — but how will it change the game? Will it enable major disruption of next-generation IP communications services? Will it usher in a paradigm shift, moving communications services to a model of value-added features for larger services, such as social media sites, and thereby eliminate the need for communications as a standalone service for direct revenue? Or even if standalone communications services continue to exist, will WebRTC make the new global ecosystem simpler?

To answer these questions, Alcatel-Lucent conducted a study of two basic WebRTC communications systems architecture models: a WebRTC islands model and a WebRTC interconnect model. The results may surprise you.

EXECUTIVE SUMMARY

Web Real-Time Communication (WebRTC) is a game-changing technology. At the very least, it lowers the barrier for new entrants to get into the communications business by making a multi-device client easy to build once and run everywhere, including when integrated into websites and web apps. This can lead to a further explosion of communications islands into the market.

The critical question is whether WebRTC will enable major disruption of next-generation IP communications services: voice, video, instant messaging, file transfer, etc., all over IP. Will it usher in a paradigm shift, moving communications services to a model of value-added features for larger services, such as social media sites, and thereby eliminate the need for communications as a standalone service for direct revenue? Or even if standalone communications services continue to exist, will WebRTC make the new global ecosystem simpler?

Can this new paradigm make interoperability between communications services unnecessary? Are islands of proprietary, WebRTC-enabled IP communications services sufficient to replace today's globally interconnected, standards-based Public Switched Telephone Network/Public Land Mobile Network (PSTN/PLMN)?

And will the URL replace the phone number?

WebRTC enthusiasts envision a future where the answers to all these questions is "yes".

This view of the future can be illustrated with a simple use case. Imagine that Alice wants to call Bob. With WebRTC, Alice will go to Bob's communications service provider (CSP) URL in her browser, click on the link for making calls, and Bob's system will set up the call between Alice and Bob. Unlike today, where Alice's CSP would set up the call with Bob's CSP, only one provider is required for a call. That provider is the one serving the called party. In this case, it is Bob's provider. Furthermore, Bob's provider does not require Alice to be a subscriber. Bob's communications system can be:

- A standalone communications service provider such as AT&T or Google Talk
- An application service provider such as Facebook or LinkedIn®
- A business, for example, Bob's employer
- A private communications application, which could be purchased from a third party, built from scratch or leverage open source code such as Asterisk®

For a simple point-to-point call, it is obvious that interoperability between two communications systems is unnecessary. In fact, it appears that this new paradigm could radically simplify individual communications systems as well as the global ecosystem.

So, the heart of the question is whether this WebRTC proprietary islands global ecosystem vision is a truly viable PTSN/PLMN replacement.

To answer this question, Alcatel-Lucent conducted a study. First, we identified the requirements for a next-generation, global, IP communications ecosystem that would replace today's PSTN/PLMN. Then, we looked at two basic WebRTC communications systems architecture models: a WebRTC islands model and a WebRTC interconnect model.

The first model assumes that only one communications system is used at any time to support a session. The second model assumes that the originating party's CSP sets up the session with the terminating party's CSP using a common Network-to-Network Interface (NNI).

We conducted the study by examining and comparing the requirements compliance for a future ecosystem of independent WebRTC proprietary islands (as defined today[1]) to a future ecosystem of WebRTC IP Multimedia Subsystem (IMS) systems. The study and results are described in this paper. The section Comparing the Two Models includes a table summarizing the requirements compliance of both ecosystem models.

Let's use one example to illustrate a potential issue with the WebRTC proprietary islands model. WebRTC requires the ability for the communications app to access the user's device (camera and microphone). From a trust standpoint, this can be a problem for the calling party, who has no relationship with the called party's CSP.

Consider Alice calling Bob. The risk is small if Bob's CSP is known and trusted by Alice. But if Bob has built his own system or signed up with HackersTalk.com, Alice may be concerned about giving Bob's system permission to access her device. This is similar to the danger of clicking on unknown web links.

This problem can be minimized with the interconnect model. Alice signs up with a provider of her choice. This provider will be the only one able to access her device and this provider will set up sessions on her behalf. The strengths and weaknesses of the two models are discussed in this paper.

Today, two billion+ out of seven billion people are Internet-connected, and there are six billion mobile subscriptions in the world[2]. This simple fact makes the WebRTC proprietary islands model incapable of fully replacing the PSTN at this time. However, the transition away from the PSTN has already begun and the reach of the Internet continues to grow quickly.

While an IMS-based interconnect model is not the only way to address the needs of a global communications ecosystem, alternatives based on proprietary islands require substantial work in both standards and regulatory bodies. Realistically, this will take some time. However, Internet players do not wait for all the standards to be ratified before implementing and deploying. So, an ad-hoc evolution and transition is taking place now even though full replacement is not possible.

As long as the PSTN/PLMN exists as a fallback and over-the-top (OTT) communications apps are exempt from regulations, WebRTC has the potential to play a major role in accelerating the value destruction of communications and taking away large market share from Telcos. Action must be taken to incorporate IMS with WebRTC immediately to reap the benefits as well as to mitigate the coming threats. WebRTC with IMS meets all the requirements for a PSTN/PLMN replacement. If Telcos move now, efforts for a full replacement alternative will be unnecessary and redundant.

[1] Future refinement of the WebRTC island paradigm may address some of the issues described in this paper. However, rather than speculating on potential future enhancements of WebRTC islands, we focus on only the current state and on standards activities known to be in progress.
[2] Sources: http://www.slideshare.net/kleinerperkins/kpcb-internet-trends-2013 and http://www.3gpp.org/6-Billion-Growing.

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INTRODUCTION

WebRTC is a game-changing technology. At the very least, it lowers the barrier for new entrants to get into the communications business by making a multi-device client easy to build once and run everywhere, including when integrated into websites and web apps. This can lead to a further explosion of communications islands into the market.

This paper examines the potential of WebRTC for disruption. Will there be a paradigm shift that moves communications services to a model of value-added features for larger services, such as social media sites? Will this shift replace communications as a stand-alone service for direct revenue?

Can this new paradigm make interoperability between communications services offerings unnecessary? Are islands of proprietary, communications services sufficient to replace today's globally interconnected, standards-based PSTN/PLMN?

Will the URL replace the phone number?

This paper answers these questions by comparing the strengths and weaknesses of a global ecosystem of WebRTC IMS systems with a global ecosystem of WebRTC proprietary islands. In the end, there will inevitably be islands for at least niche services. The question is whether an islands-only approach is all that is needed.

REQUIREMENTS OF A NEXT-GENERATION IP COMMUNICATIONS GLOBAL ECOSYSTEM

Before examining the strengths and weaknesses of the different network architecture models, we first need to outline the requirements of a next-generation IP communications global ecosystem. Given that today's global PSTN and PLMN ecosystem has made a social contract of universal reach possible and expected (for example, reachability is a regulatory requirement in some countries), the next-generation IP communications ecosystem must do no less.

In the following sections, two lists are provided. The first is the list of absolute requirements necessary to support truly universal communications. The second is a list of requirements that enable a better user experience, but it can be argued whether or not they are must-haves.

Both types of requirements apply to consumer-to-consumer (C2C), consumer-to-business (C2B), business-to-consumer (B2C) and business-to-business (B2B) use cases.

There is no implication that every communications system must adhere to the absolute requirements. These requirements apply to the global ecosystem, meaning that there is a need for everyone to have at least one option to meet each requirement.

Absolute requirements

A viable global ecosystem for next-generation IP communications must meet the following requirements.

Universal IP reachability

For universal IP reachability, communications services must be available to anyone from anywhere. People must be universally reachable through voice, video and messaging.

Reliable calling party identification

Reliable calling party identification requires that users and their CSPs be able to confidently identify callers and optionally screen incoming calls. Unwanted calls are the unavoidable consequence of universal reachability. Users and the CSPs, on the subscribers' behalf, must be provided with reliable information regarding the identity of the calling party and with the ability to proactively filter incoming calls.

Access to trusted service

Users need to be able to confidently call another party and not worry about a nefarious serving communications system causing possible harm on their device. Web users today must be leery of clicking on unknown links, and WebRTC links are a new type of web link.

Users may have concerns if they need to use the clients from each called party's system, allowing many different and sometimes unfamiliar systems access to their device(s)[3].

Interworking with PSTN/PLMN endpoints

There must be communications providers in the ecosystem that are capable of interconnecting with PSTN/PLMN endpoints.

Many users will be served by only the PSTN/PLMN for some time. Users must be able to originate calls to and terminate calls from these PSTN/PLMN endpoints. As a result, WebRTC providers must be able to support E.164 numbers for their subscribers.

Service/feature availability

Users must have access to the services offered by their chosen provider regardless of call origin/destination. Of course, some services will not be compatible with another party's provider, so those services may not be usable in those instances.

Ad-hoc group communications

When offered as part of their service plan, users must be able to include any party with compatible services in group communications. Voice is a minimum requirement. Video, messaging and sharing should also be available when supported.

Charging and billing framework

Providers must have the flexibility to support multiple charging and billing models.

To facilitate a variety of business models, the ecosystem must support the capability to exchange sufficient call detail information for flexible charging arrangements, such as calling party pays, called party pays, bill and keep, etc., according to inter-provider and subscriber agreements.

[3] The "trapezoidal" model, in which a user subscribes to a service that interconnects with other services on their behalf, avoids this issue because the user inherently trusts the CSP that he/she signed up with. There is nothing inherent in WebRTC that prevents the use of such a model.

Innovation

Providers must be capable of unilaterally offering new services without impacting universal basic communications or relying on a standardization process.

Emergency services

Emergency services must be universally available.

A next-generation ecosystem will be regulated. Regulatory services, including emergency services (voice, video, messaging) and access priority are required. Access providers will be required to provide location and priority. Related to this, public safety systems must be supported by the new global ecosystem.

Lawful intercept

A next-generation ecosystem will be regulated. Regulatory services, including lawful intercept, must be supported.

Requirements for an improved user experience

A global ecosystem for next-generation IP communications should meet the following requirements for an improved user experience.

Address consolidation

Users should be able to designate a few "primary" contact points.

Users will have many URLs, and they must be able to route from one to another. Google™ Voice™ is an example of a service that provides this capability for today's PSTN/PLMN.

Address portability

Users should be able to port their contact address from one provider to another. This is a regulatory requirement for E.164 numbers and a desirable feature for private domain URLs.

Users should be able to change providers without needing to change the contact address, especially if they are using a "vanity" domain or a non-communications service provider-related domain[4] (for example, Alice.Smith@ieee.com).

Subscriber choice of provider

Users should be able to select the communications provider they want to use for originating and terminating communications services.

Calling parties should not be forced to use a provider that is not of their choosing. A calling party may prefer to use her own provider to set up a communications session with a called party rather than using the called party's provider.

Contact addresses, especially URLs, from unknown sources or strangers may be viewed with suspicion. This is true from the perspective of an individual subscriber as well as from the perspective of a business, which can get a large number of "junk" URLs from customers.

[4] It is not expected that URLs will be portable from one enterprise domain to another. For example, j.doe@ijk.com will not be supported by xyz.com.

Outgoing call restrictions

Users should be able to restrict outgoing calls.

In both the consumer and enterprise markets today, there are features available to allow outgoing calls to only a select set of numbers or to forbid outgoing calls to specific numbers. These should also be provided in the WebRTC ecosystem.

Session transfer

Users should be able to transfer an existing call from one endpoint to another regardless of which provider is supporting either endpoint.

Private call transfer should be possible, that is, the party being transferred should not be told the target address of the transfer.

Session forwarding

Users should be able to forward an incoming call to an endpoint on a system other than the original called number’s terminating system.

Private call forwarding should be possible, that is, the party being forwarded should not be told the target address of the forwarded session.

Standalone communications devices

Users should have options to communicate using standalone devices, that is, users should not be restricted to needing to open a browser on a PC or tablet at home to make or receive calls.

Optimal use of wireless spectrum

Wireless spectrum is a limited, valuable resource, and many studies project serious spectrum challenges to meet future demands. Therefore, an architecture that optimally uses wireless spectrum is preferred from a network resources perspective.

This architecture includes optimal signaling and bearer plane design in addition to the CSP being mobile aware. For example, systems in which conference mixing is performed in the network with only a single stream to the endpoint are more bandwidth efficient than those relying on multiple streams to each endpoint. Therefore, a client-server design is better than a fully-interconnected peer-to-peer mesh. This applies to not only audio and video conferencing but to multi-party file transfer and instant messaging with payloads, such as images.

CSPs that are mobile aware also know to use wireless codecs (coders-decoders) for maximum efficiency (for example, Adaptive Multi-Rate [AMR] instead of G.711) and to request Quality of Service (QoS) when available.

Regulatory issues

In the next-generation communications ecosystem, an interesting challenge arises for government regulators. That challenge is to determine which service providers (CSPs and/or ISPs) should be subject to regulation. For example, in an Internet-based communications world, is lawful intercept the responsibility of the CSP or the access provider? New service provider profiles will need to be created and regulations re-thought in light of the new reality.

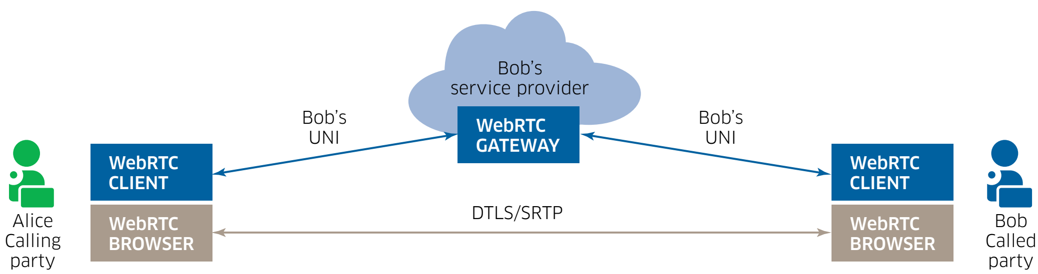
WebRTC ISLANDS ECOSYSTEM MODEL

Network architecture

In the WebRTC islands ecosystem model, there is only one standalone system that supports a communications session — that system is the one providing service for the called party.

Figure 1 shows the WebRTC islands ecosystem model.

Figure 1. WebRTC islands ecosystem model



The originating party, Alice, browses to the communications website of the terminating party, Bob, then clicks the appropriate link to launch the WebRTC client and make the call to Bob. Bob's CSP handles both the originating and terminating legs of the call. Alice's CSP is not involved. Only the terminating party's CSP is used for call-related features. Alice does not need to be a subscriber of Bob's CSP.

Issues related to basic call scenarios

Table 1 shows the issues associated with basic call scenarios for this model. Although the scenarios discuss C2C situations, the same issues and questions also apply to B2C and B2B situations.

Table 1. Issues related to basic call scenarios with the WebRTC islands ecosystem model

ISSUE #	ISSUE DESCRIPTION
1	Alice is restricted to using only those communications features that Bob's CSP offers. Therefore, Alice's communications features will be different depending on which CSP her called party is using.
2	An ID management mechanism must be provided to allow Bob's CSP to reliably determine who Alice is. Bob may have Alice on a do-not-call list. Mechanisms that allow the called party to reliably identify the calling party are being discussed in standards bodies. These mechanisms could be leveraged to support call filtering by the called party's CSP.
3	Flexible charging arrangements should be supported. For example, if Bob's CSP requires payment for terminating calls to its subscribers, Alice must subscribe to a payment service that is accepted by Bob's CSP5. Charging for terminating calls can be one way to mitigate large volumes of spam or other unwanted calls.
4	Most countries will require the ability to support lawful intercept on Alice's calls if she is the target of a law enforcement investigation. However, the fact that Alice does not use her own CSP to originate calls complicates lawful intercept. In the WebRTC islands model, data stream intercept by Alice's access provider may be the only option for law enforcement.

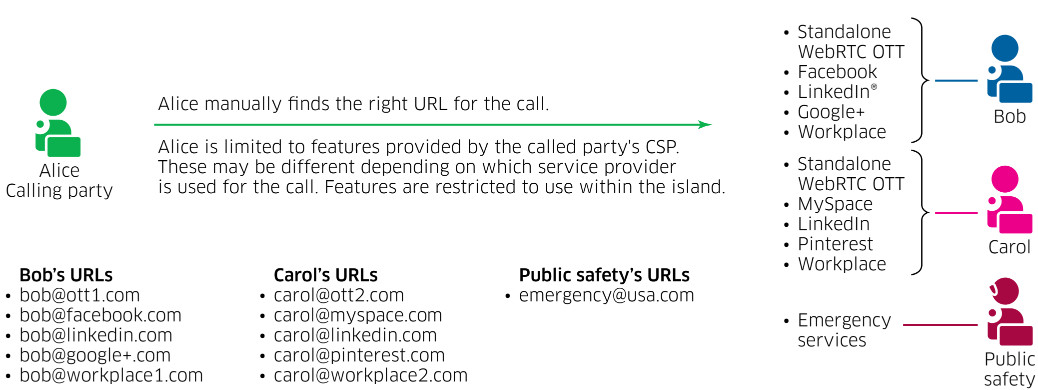
[5] This calling-party-pays model is especially useful to support unsolicited business calls (for which Bob is unwilling to pay). A micropayment infrastructure is also believed to be an effective deterrent for spam or junk calling because mass calling becomes prohibitively expensive for the spammer.

ISSUE #	ISSUE DESCRIPTION
5	It is not clear how Alice can be assured that Bob's CSP can be trusted to do no harm. If Bob's CSP is a large Telco such as AT&T or a large OTT provider, Alice might feel safe. However, if Bob's CSP is NoName.com or HackersRUs.com, Alice would not want to call Bob if it requires using Bob's system. No careful person would carelessly click on any link when web browsing. The same would be true when using WebRTC to call someone.
6	When calling Bob, Alice is presented with the user interface from Bob's CSP, which may be unfamiliar and difficult for her to navigate. Even worse, if Bob lives in another country that uses a different primary language from Alice, Bob's communications portal will most likely be in that language. If Alice cannot read that language, this could be a problem. This could be a general issue for international calling, although multilingual web sites exist today[6].
7	Just as Facebook is banned in some countries, it is likely that the island model will not be allowed in all countries[7] – either for calling the citizens of the country or for those citizens to call out of the country. There will be situations where governments and ISPs will choose to block certain communications services. These restrictions could extend to WebRTC, making the island model less able to be global.
8	Given that no originating CSP is involved in the call, how can outgoing call restrictions be supported[8]? For example, how can a child be restricted to calling only five phone numbers or be restricted from calling specific WebRTC island sites? The same issue applies for enterprises and their employees.

Interaction with multiple contacts

Now, let's take a step back to see how Alice relates to all her communications contacts and needs in the WebRTC islands ecosystem model (see Figure 2).

Figure 2. Friends, family and emergency services



Given that Alice's contacts are served by a variety of (possibly non-overlapping) CSPs, issues arise around how Alice simultaneously interacts with multiple contacts — a critical feature for an increasingly socially networked world. It is assumed that Alice's address book is capable of holding URLs for each of her contacts and allowing her to connect with them through a simple touch or mouse click. However, more complex interactions are challenging.

[6] The ATIS ORCA project would allow for Alice to have a single communications user interface app that can interface to multiple WebRTC CSPs if the CSPs build ATIS ORCA-compliant transport libraries for their systems. This would allow Alice to use her single app to interface with international WebRTC CSPs.
[7] In the past, the USSR used to block outgoing international calls. Today, there are still some countries with similar restrictions.
[8] If there is market demand, services could emerge to support call origination restrictions. Such a service would likely be provided in conjunction with a general proxy/firewall service that restricts access to specific web sites.

Issues related to complex call scenarios

Table 2 shows issues related to more complex call scenarios for the WebRTC islands model. Although the scenarios discuss C2C situations, the same issues and questions also apply to B2C and B2B situations.

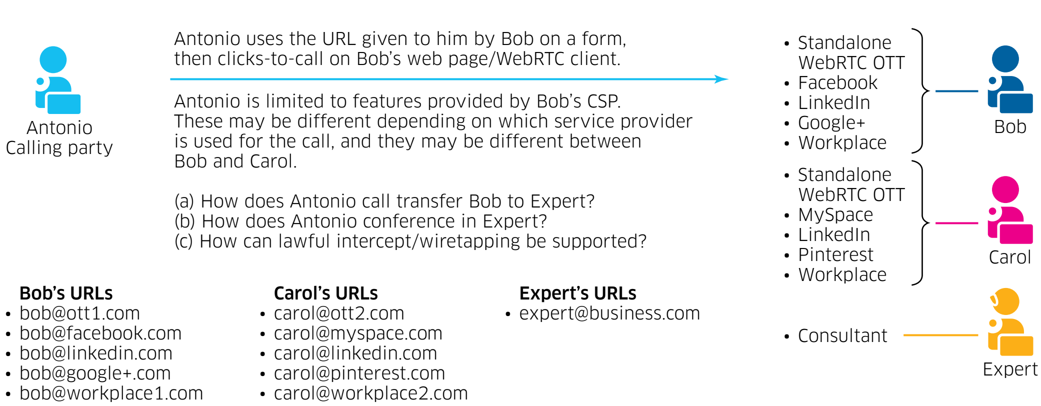
Table 2. Issues related to complex call scenarios with the WebRTC islands ecosystem model

ISSUE #	ISSUE DESCRIPTION
1	How can call transfer be supported if Alice wants to transfer the call with Bob to Carol? This would require manual re-direct or inter-island interaction to address privacy issues in cases where Bob and Carol should not have access to each other's contact information.
2	How can call forwarding be supported if Alice wants to forward calls from one CSP to another? This would require manual re-direct or inter-island interaction to address privacy in cases where the caller should not have access to the callee's forwarding information.
3	How can ad-hoc conferencing be supported if Alice wants to add Carol after starting a session with Bob? This would require inter-island interaction or manual re-direct to another standalone conferencing service.
4	How can ad-hoc group chat with participants from different islands be supported? This would require inter-island interaction or manual re-direct to another standalone conferencing service.
5	Emergency services calls must be properly routed to the right Public Safety Answering Point (PSAP). This is being discussed in standards committees.
6	How can Alice call a legacy PSTN/PLMN phone? Here are three methods. • Legacy telecom service providers must provide a WebRTC gateway (GW) and portal for their PSTN subscribers. An Electronic Number Mapping System/Domain Name Server (ENUM/DNS) service is needed that allows Alice to first discover Bob's provider based on his E.164 number. Then, Alice would use that provider's portal to call Bob. • An independent provider could handle calls to the PSTN/PLMN using a WebRTC interface provided the needed charging/billing infrastructure is in place for the calling party. • Alice's CSP could discover Bob's gateway portal and also set up the call for Alice. This is the WebRTC interconnect model described in the next section.
7	How can a legacy PSTN/PLMN phone call Alice? If she is to be reachable from the PSTN, Alice's WebRTC island provider must have a gateway to the PSTN and support an E.164 number for Alice. This is equivalent to the SkypeIn™ service that exists today.

B2C and B2B use cases

Figures 3 and 4 show the B2C and B2B use cases for the WebRTC islands model. In these scenarios, Antonio has a business. Bob and Carol are Antonio's customers, and the expert is a business partner.

Figure 3. WebRTC islands model for B2C

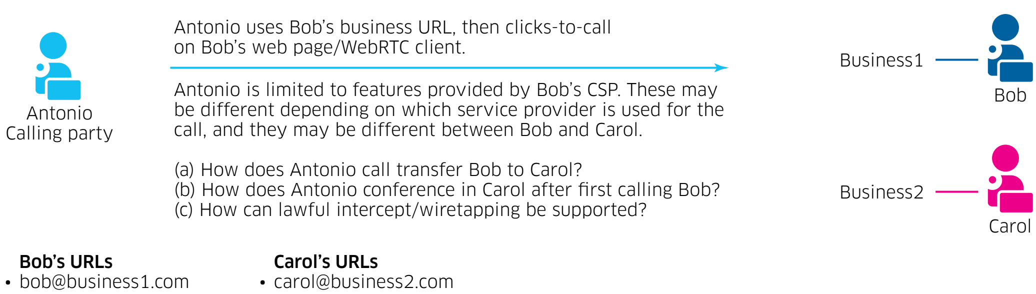


For a B2C scenario, the fact that Antonio's business is reliant on the WebRTC island of the customer (Bob) for communications features is a serious problem. If Antonio needs to conference in an expert from another WebRTC island or to transfer Bob to the expert on another island, there is no satisfactory solution using the WebRTC islands model.

An agreed-upon, standards-based interworking mechanism that supports conferencing and/or call transfer between islands is needed. Furthermore, private call transfer is often required: the transfer must be made without disclosing to the party being transferred the address of the transfer target (that is, the expert).

The same is true for B2B scenarios, as shown in Figure 4.

Figure 4. WebRTC islands model for B2B



The WebRTC islands model can address many communications needs in a new, arguably easier-to-use paradigm, leveraging the URL address. However, there are also many gaps and issues. Examples include:

- Legacy PSTN/PLMN interworking
- User trust concerns due to the fact that each person will have no choice but to use many different and sometimes unknown WebRTC islands for each individual or business he or she needs to call
- Per-country restrictions and regulations
- The need for charging flexibility

While many issues can be addressed in a multitude of ways, it is unclear how all issues can be resolved fully and in such a way as to support global and interoperable communications. It is obvious that many solutions will require standardization.

WebRTC INTERCONNECT ECOSYSTEM MODEL

Overview

An alternative to the WebRTC islands model is a WebRTC interconnect model. In this model, communications systems using WebRTC clients interoperate with each other using a common NNI.

In the WebRTC interconnect model, calls are handled by two communications providers: an originating CSP that handles all aspects of communications on behalf of the calling party, and a terminating CSP that handles calls on behalf of the called party.

The strength of this model is that it addresses many of the shortcomings of the WebRTC islands model. For example, the WebRTC interconnect model can shield the caller from direct interaction with unknown providers while supplying a consistent user interface and feature set.

The potential, theoretical, weakness of this model is that it requires service providers to agree to use a common NNI or provide pair-wise gateways with those providers with whom interworking is to be supported — and the latter approach does not scale well. However, 3GPP defines a common NNI for IMS service providers, so practically speaking, this is not a weakness for a WebRTC interconnect model that leverages IMS.

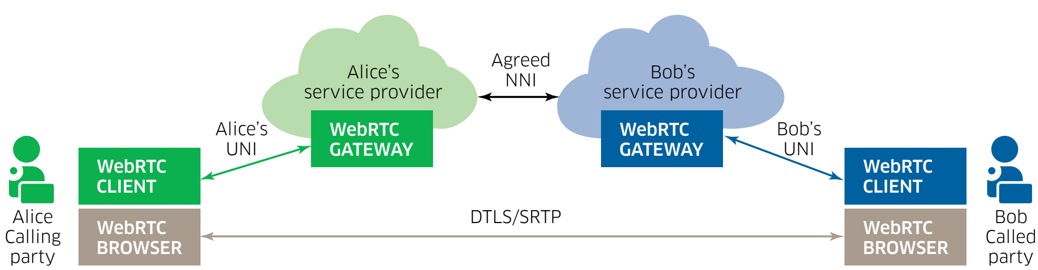
Another potential weakness is that a standardized interconnect model stifles or slows innovation. Standards, in general, stifle innovation. However, for IMS, the standards are mature and are already being deployed. They support all the foreseeable types of communications services. As such, they provide the ideal foundation for a global WebRTC ecosystem.

In addition, Application Programming Interfaces (APIs) for IMS are also defined and being deployed. These APIs and the already available foundational services are building blocks that enable rapid innovation and development of new features and applications, and integration of communications services into existing applications.

Network architecture

Figure 5 shows the network architecture of the WebRTC interconnect ecosystem model.

Figure 5. WebRTC interconnect ecosystem model



With this model, Alice does not need to worry about a multitude of CSPs — with whom she does not have a relationship — accessing her device(s). Her CSP sets up and negotiates calls on her behalf to the systems of the called parties. This is a valuable service.

For basic calls, Alice's WebRTC interconnect CSP originates the call to Bob's WebRTC interconnect CSP. Alice has access to her originating service feature set regardless of who she calls (Bob, Carol, etc.) because the feature set comes from her CSP.

For example, Alice's WebRTC CSP can support ad-hoc conferencing of Carol — who is with yet another WebRTC CSP — into an existing call between Alice and Bob. Or Alice's WebRTC CSP can support Alice transferring Bob to Carol in a private manner. This is possible because these features have been standardized and the interworking procedures agreed upon.

Alice's CSP also handles ID management and supports various charging and billing arrangements with Bob's CSP.

The WebRTC interconnect model meets all of the next-generation IP communications ecosystem requirements.

WebRTC IMS ECOSYSTEM

Overview

A WebRTC IMS ecosystem can include IMS systems using WebRTC operating in island mode and in interconnect mode. Using IMS as the communications back end with WebRTC provides the flexibility and advantages of both models.

Scenarios easily supported by a WebRTC IMS ecosystem are calls to and from the PSTN/PLMN, QoS, lawful intercept and all other regulatory services. These are defined for the IMS ecosystem today and will extend to the use of WebRTC clients with IMS. Innovative first-party and third-party mashup applications can also be created quickly on an IMS system using WebRTC technologies and APIs.

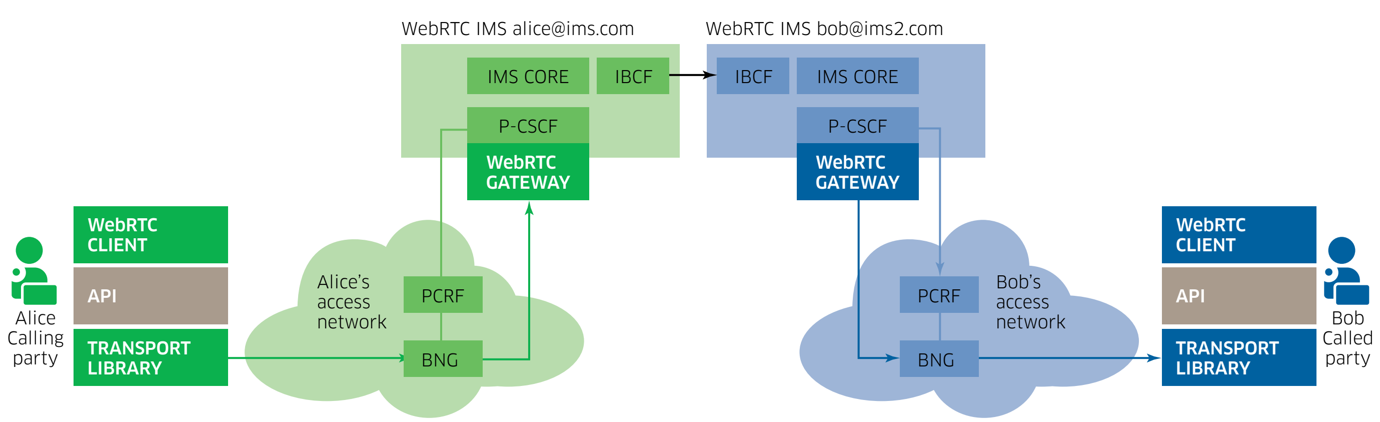
The combination of WebRTC and IMS is very powerful. IMS service providers should exploit this combination early to both gain the benefits from WebRTC and grab market share even for island model solutions. An IMS service provider has a single next-generation IP communications system that can be used as a platform for many WebRTC solutions. This is also an advantage over a collection of siloed WebRTC solutions that replicate the same capabilities multiple times.

We recommend that Telcos start deploying WebRTC for both the enterprise and consumer markets. In the enterprise market, wholesaling IMS communications services to e-commerce sites, social networking and dating sites, financial and healthcare sites, gaming platform providers, etc. are new Internet market channels with high potential. In the consumer space, we recommend starting with a web extension solution for IMS that includes support for calls to and from the PSTN/PLMN. We provide five use cases to illustrate this solution. The use cases leverage both the interconnect and islands models. (Appendix A describes additional scenarios.)

WebRTC IMS use cases

In the first use case, Alice uses her WebRTC IMS provider to call Bob at his WebRTC IMS provider (see Figure 6).

Figure 6. Alice uses her WebRTC IMS provider to call Bob at his WebRTC IMS provider

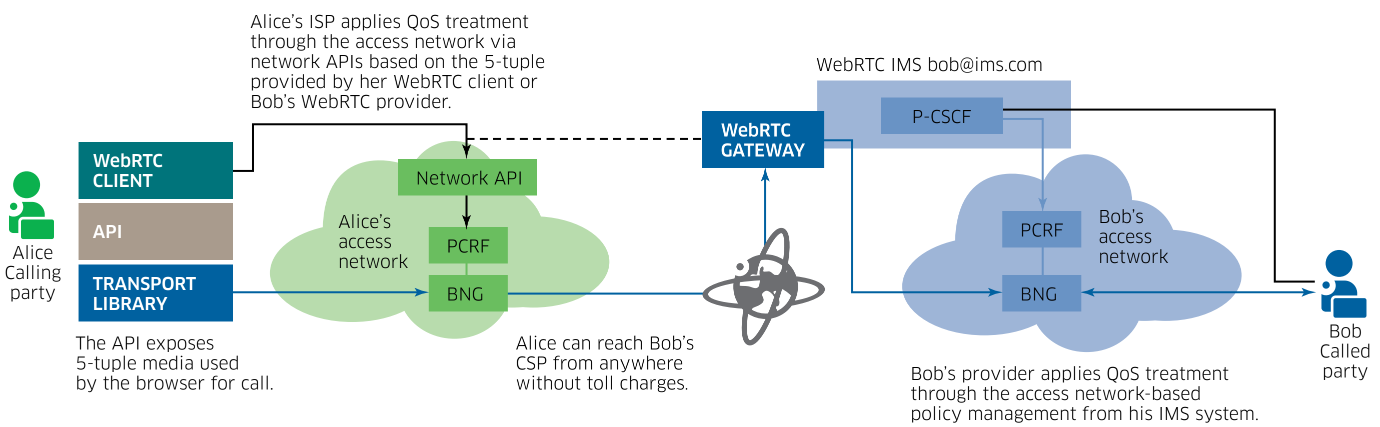


The procedure is as follows.

1. Alice uses her WebRTC IMS portal to initiate a call to Bob.
2. Alice's provider finds Bob using standard IMS procedures.
3. Alice gets QoS on the access link if it is part of her package and she is using an access network managed by her IMS provider.
4. Bob gets QoS on the access link if it is in his package and he is using an access network managed by his IMS provider.

In the next use case, Alice uses Bob's IMS WebRTC portal to call Bob using Bob's URL (see Figure 7). This island scenario allows Bob to extend his IMS advanced communications services to friends and family who may not yet be IMS subscribers.

Figure 7. Alice uses Bob's IMS WebRTC portal to call Bob

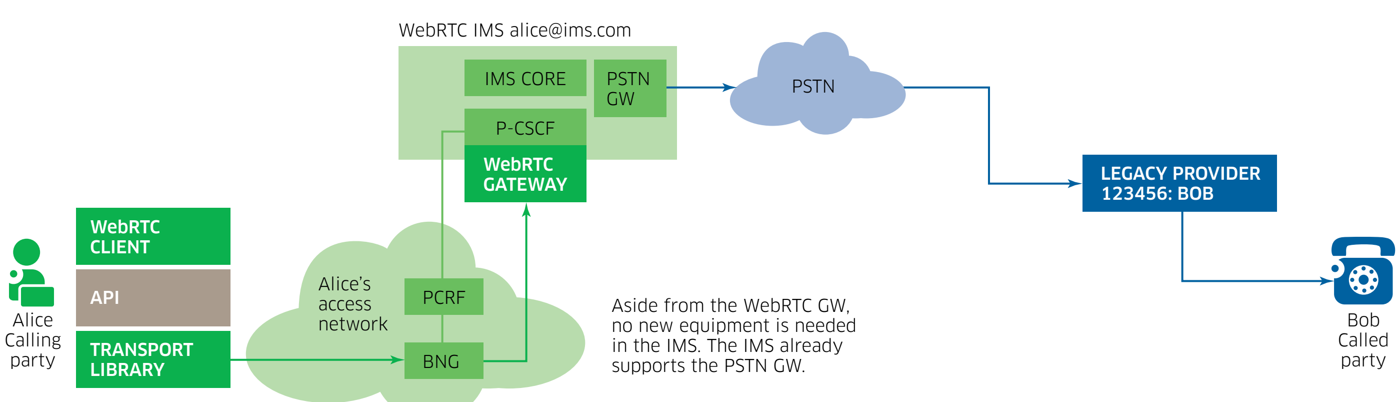


The procedure is as follows.

1. Alice uses her access provider to connect to Bob's IMS WebRTC portal. Optionally, Alice's access provider may provide QoS at the request of her client or Bob's WebRTC gateway.
2. Bob's provider gives Alice originating services and sets up the call to Bob.
3. Bob's provider terminates the call to Bob and gives him QoS.

Another scenario is for Alice to use her WebRTC IMS provider to call Bob on the PSTN/PLMN (see Figure 8).

Figure 8. Alice uses her WebRTC IMS provider to call Bob on the PSTN/PLMN

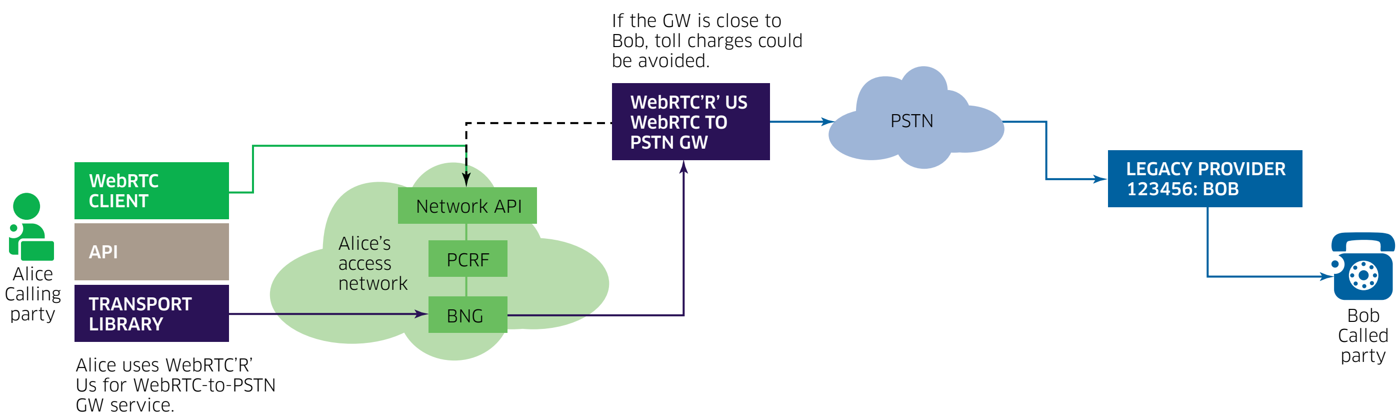


The procedure is as follows.

1. Alice makes a call to Bob using Bob's E.164 number on her WebRTC IMS client.
2. Alice's IMS provider uses standard IMS procedures to call Bob on the PSTN.

Alice can also use an independent WebRTC IMS provider to call Bob on the PSTN/PLMN (see Figure 9).

Figure 9. Alice uses an independent WebRTC IMS provider to call Bob on the PSTN/PLMN



The procedure is as follows.

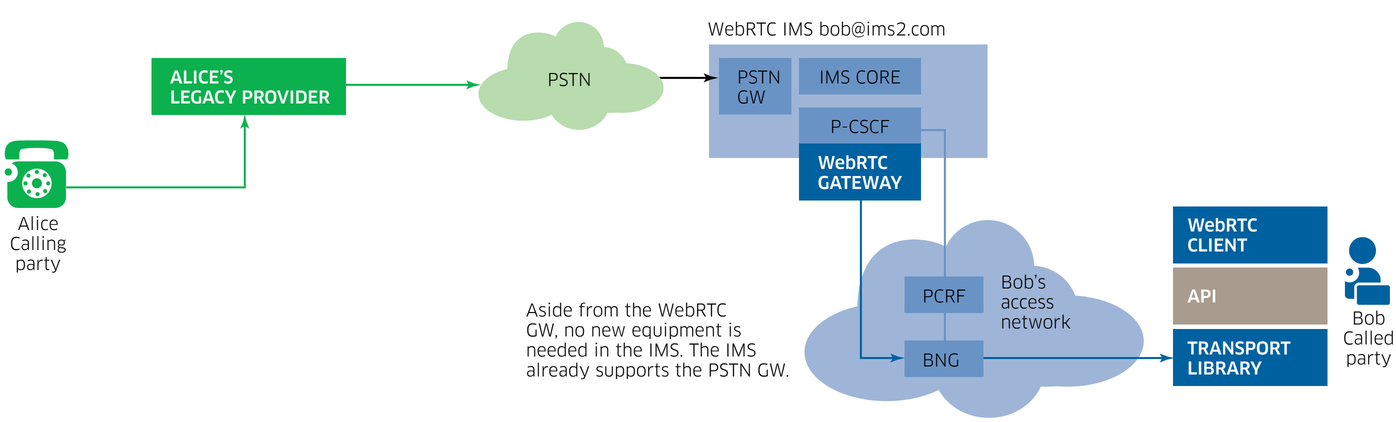
1. Alice uses Bob's WebRTC IMS portal or an independent IMS provider to make a call to Bob using Bob's E.164 number.
2. Bob's IMS provider or an independent IMS provider uses standard IMS procedures to call Bob on the PSTN.

   Called or calling party pay scenarios are possible.

   Note: There is a price advantage if the gateway is located close to the PSTN exchange serving Bob, making it a local call.

Another scenario is for Alice to use the PSTN to call Bob at his WebRTC IMS provider (see Figure 10).

Figure 10. Alice uses the PSTN to call Bob at his WebRTC IMS provider



The procedure is as follows.

1. From the PSTN, Alice dials Bob's number. The call is routed, using standard procedures, to the IMS PSTN gateway provided by Bob's IMS provider.
   Note: Alice pays any international/long distance charges.
2. Bob's IMS provider terminates the call to Bob's device.

COMPARING THE TWO MODELS

Table 3 summarizes the ability of the current WebRTC proprietary islands and WebRTC IMS models to support the requirements of a next-generation IP communications, global ecosystem.

Table 3. Requirements support of WebRTC islands and WebRTC IMS

REQT #	REQUIREMENT DESCRIPTION	WebRTC PROPRIETARY ISLANDS	WebRTC IMS ECOSYSTEM	COMMENT
1	Universal IP reachability	Y	Y	
2	Reliable calling party identification	No established convention	Y	Defined in 3GPP for IMS
3	Access to trusted service	Partial	Y	Inherent in IMS model
4	Interworking with PSTN/ PLMN	Partial	Y	Defined in 3GPP for IMS
5	Calling party originating feature availability	N	Y	Defined in standards for IMS
6	Ad-hoc group communications	No established convention	Y	Defined in standards for IMS
7	Charging and billing framework	No established convention	Y	Defined in 3GPP for IMS
8	Innovation	Y	Y	IMS API standards
9	Emergency services and access priority	Require government action	Y[9]	Defined in 3GPP for IMS
10	Lawful intercept	No established convention and requires government action	Y	Defined in 3GPP for IMS
11	Address consolidation	Partial	Y	
12	Address portability	N	Partial	Defined in 3GPP for IMS
13	Subscriber choice of (origin) provider	N	Y	Inherent in IMS model
14	Outgoing call restrictions	No established convention	Y	
15	Session transfer (private)	N	Y	Defined in standards for IMS
16	Session forwarding (private)	N	Y	Defined in standards for IMS
17	Standalone communications device	Partial[10]	Y	
18	Optimal use of wireless spectrum	Architecture dependent[11]	Y	Inherent in IMS model

[9] Solution alternatives for retrieving location information from the device are being investigated.
[10] In theory, these devices could be productized for WebRTC islands, but they do not exist today. Dedicated home communications devices exist today for IMS solutions; these could serve as a base for WebRTC versions.
[11] WebRTC proprietary islands can be implemented using one of many different architecture design options. Compliancy depends on the architecture design choice of any particular island.

On a feature-by-feature basis, the WebRTC IMS ecosystem model offers more. It supports the following features that the WebRTC proprietary islands model does not:

- Calling party originating feature availability
- Address portability
- Subscriber choice of (origin) provider
- Session transfer (private)
- Session forwarding (private)

In addition, the WebRTC IMS ecosystem model provides full support for the following features while the WebRTC proprietary islands model offers only partial support:

- Access to trusted service
- Interworking with PSTN/PLMN
- Address consolidation
- Standalone communications device

The WebRTC IMS model also fully supports the following features, for which the WebRTC islands model has no established convention:

- Reliable calling-party identification
- Ad-hoc group communications
- Charging and billing framework
- Lawful intercept
- Outgoing call restrictions

The WebRTC islands model also requires government action for emergency services and access priority as well as for lawful intercept. And this model's capability to make optimal use of the wireless spectrum is architecture dependent.

By contrast, the WebRTC IMS model offers full support of all compared features except address portability, for which it offers partial support.

CONCLUSION

The combination of WebRTC and IMS is very powerful. It meets all the requirements of a next-generation communications ecosystem. It can run in standardized interconnect mode and in island mode. It can also support both URL and E.164 addressing paradigms, the latter of which will continue to be attractive — and necessary — for subscribers for many years to come. The transition away from the PSTN will take time, so interoperability with legacy systems will be required. IMS can naturally provide for that interoperability need.

Today, only two billion people out of seven billion people are Internet-connected. At the same time, there are six billion mobile subscriptions. These simple facts make the WebRTC proprietary islands model incapable of fully replacing the PSTN at this time. However, the transition away from the PSTN has already begun and the reach of the Internet continues to grow quickly.

IMS service providers should move now to support WebRTC and gain the benefits of expanding services into the web while leveraging their advantages over WebRTC proprietary islands. At the same time, they should grab market share for island model solutions rather than risk losing market share by doing nothing.

While an IMS-based interconnect model is not the only way to address the needs of a global communications ecosystem, alternatives based on proprietary islands require substantial work in both standards and regulatory bodies. Realistically, this work will take some time. However, Internet players do not wait for all the standards to be ratified before implementing and deploying. So, an ad-hoc evolution and transition is happening now even though full replacement is not possible.

As long as the PSTN/PLMN exists as a fallback and OTT communications apps are exempt from regulations, WebRTC has the potential to play a major role in accelerating the value destruction of communications and taking away large market share from Telcos. Action must be taken to incorporate IMS with WebRTC immediately to realize the benefits and also mitigate the coming threats. WebRTC with IMS meets all the requirements for a PSTN/PLMN replacement. If Telcos move now, efforts for a full replacement alternative will be unnecessary and redundant.

APPENDIX A: ADDITIONAL WebRTC IMS OPERATING MODELS

The figures in this appendix show four additional WebRTC IMS operating models.

Figure 11. WebRTC IMS-to-WebRTC IMS call: Traditional provider interconnect model

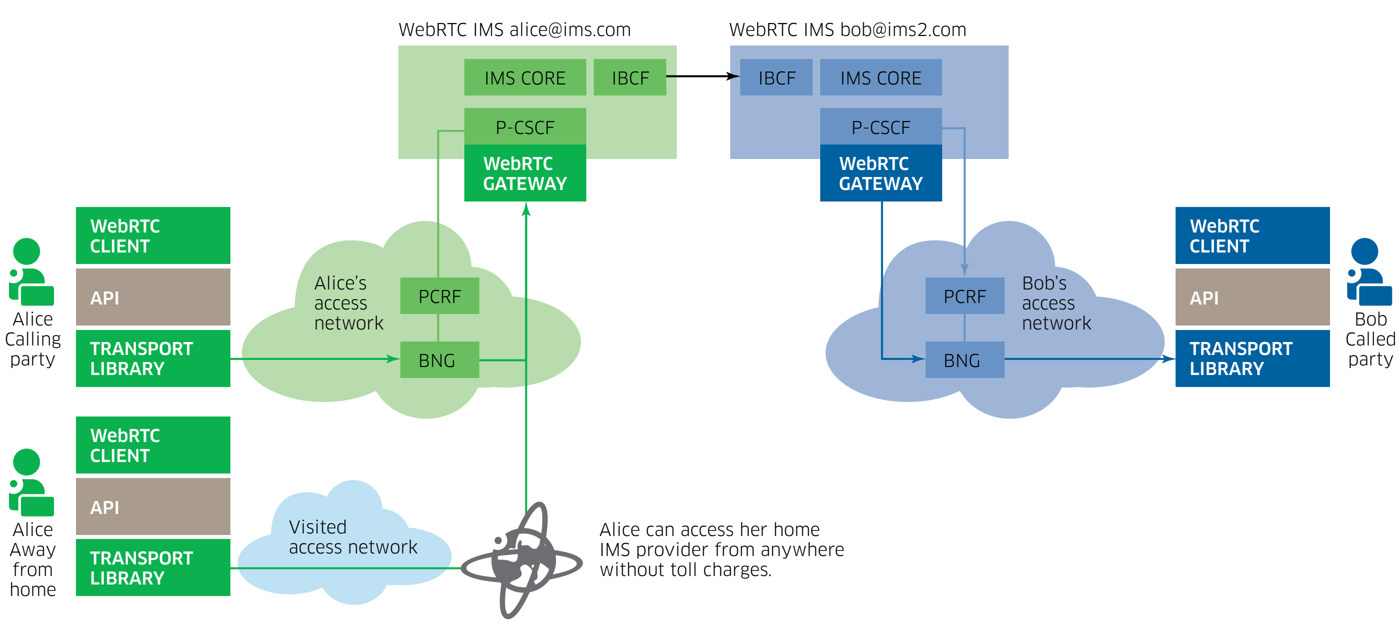


Figure 12. WebRTC IMS-to-WebRTC IMS call: Traditional provider interconnect model with separate CSP and ISP

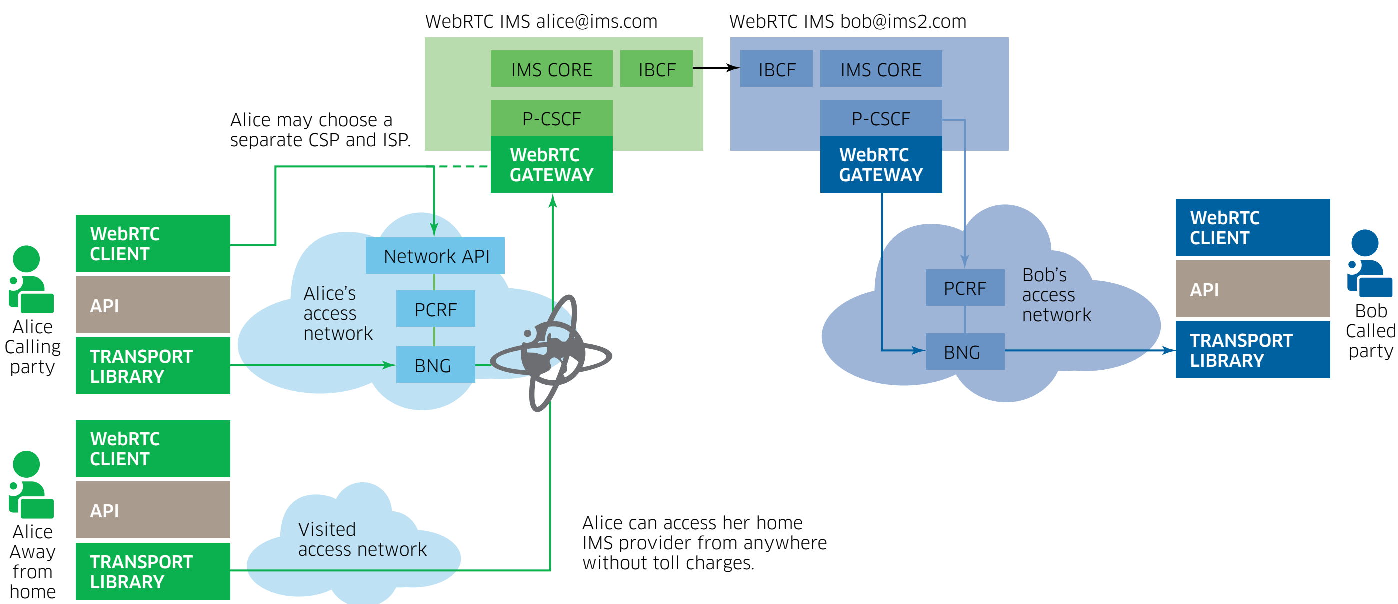


Figure 13. WebRTC IMS-to-PSTN call: Calling IMS supplies PSTN gateway

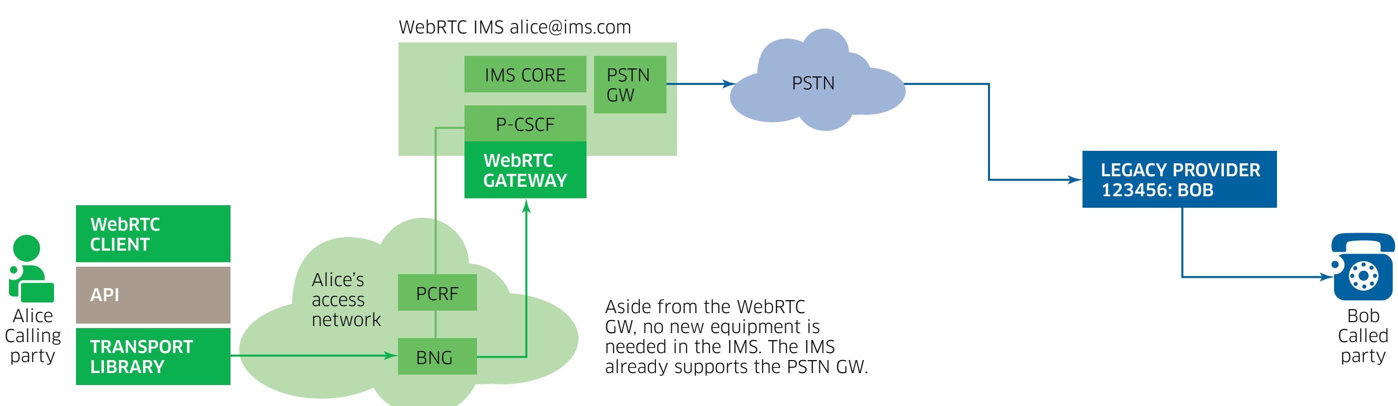
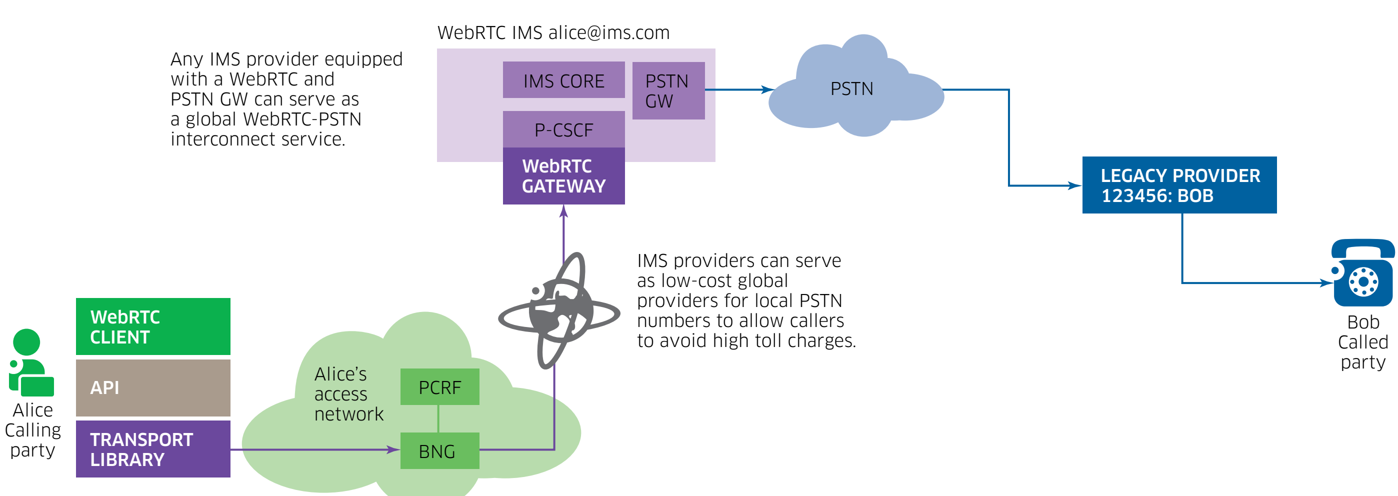


Figure 14. WebRTC IMS-to-PSTN call: IMS as global WebRTC to PSTN gateway service



ACRONYMS

3GPP	3rd Generation Partnership Project
API	application programming interface
AMR	Adaptive Multi-Rate
B2B	business to business
B2C	business to consumer
C2B	consumer to business
C2C	consumer to consumer
BNG	Broadband Network Gateway
CSP	communications service provider
DNS	Domain Name Server
DTLS	Datagram Transport Layer Security
EMS	Element Management System
ENUM	Electronic Number Mapping System
GW	gateway
IBCF	Interconnection Bearer Control Function
IMS	IP Multimedia Subsystem
ISP	Internet service provider
NNI	Network-to-Network Interface
OTT	over-the-top
PCRF	Policy Charging and Rules Function
P-CSCF	Proxy Call State Control Function
PLMN	Public Land Mobile Network
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
QoS	Quality of Service
SRTP	Secure Realtime Transport Protocol
UNI	User-to-Network Interface
WebRTC	Web Real-Time Communication

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