TELECOM INFRA PROJECT

Bring your own number (BYON) to the Metaverse based on webRTC

Use case & Architecture definition

Metaverse Ready Networks

Telecom Infra Project

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1.Background

Immersive experiences such as augmented reality (AR) and Metaverse technologies are rapidly evolving. However, today's lifestyle means that a phone call can come in at any time and needs to be answered. In this way, the integration of Coms services with these immersive technologies is almost a necessity in order to improve the user experience.

This integration can help ensure that these immersive tools are not just limited to entertainment but can lay the foundations for them to become productivity tools. Thus, the development of services such as the one described in this document should be part of the roadmap of Metaverse technologies.

2.High level Service definition – What is BYON?

Bring your own number (BYON) service aims to provide consumers with the ability to use their phone number (MSISDN) for outgoing and incoming calls from non-SIM card or non-cellular devices. Any web-based application can integrate this BYON as a feature utilizing well-known HTTP and webRTC technologies. To access the service, users are required to authenticate using their digital credentials associated with their Internet Service Provider account, for example their operator app credentials. Registration from a web endpoint is necessary to enable the service on these devices.

Outgoing calls made with BYON service will display the user's MSISDN as the caller number, ensuring that the recipient recognizes the user's phone number on their caller ID and no change is perceived by the recipient with respect to identification or audio quality.

Incoming calls introduce a unique functionality whereby multiple devices will ring simultaneously. This includes the user's smartphone and the web endpoint the user has registered from. This design enables users to conveniently answer calls from any of the registered devices, depending on their preference or accessibility. The service is compatible with various devices that support web environments, such as laptops, PCs, and XR immersive glasses.

The service specification emphasises the accessibility and flexibility of utilising one's phone number across different non-SIM card devices. By leveraging this web-based service, users can extend their communication capabilities beyond traditional SIM-based devices, enhancing their overall communication experience.

3.Value proposition

This service brings new functionalities for WebRTC applications that require to integrate the BYON API to maintain the client number in out-bound and in-bound calls from web comms service.

Out-bound calls:

- Recipient receives call as if it was from User's normal phone number
- Maintains common number through web comms service and standard subscription
- Facilitates number recognition and call-back for co-workers, partners and/or clients

In-bound calls:

- Receive phone calls simultaneously on phone subscription and web comms service
- Presents 'Occupied user' info to caller user is already in phone call
- Facilitates common multi-channel contact with co-workers, partners and/or clients

Find below the proposed high-level overview for this service:

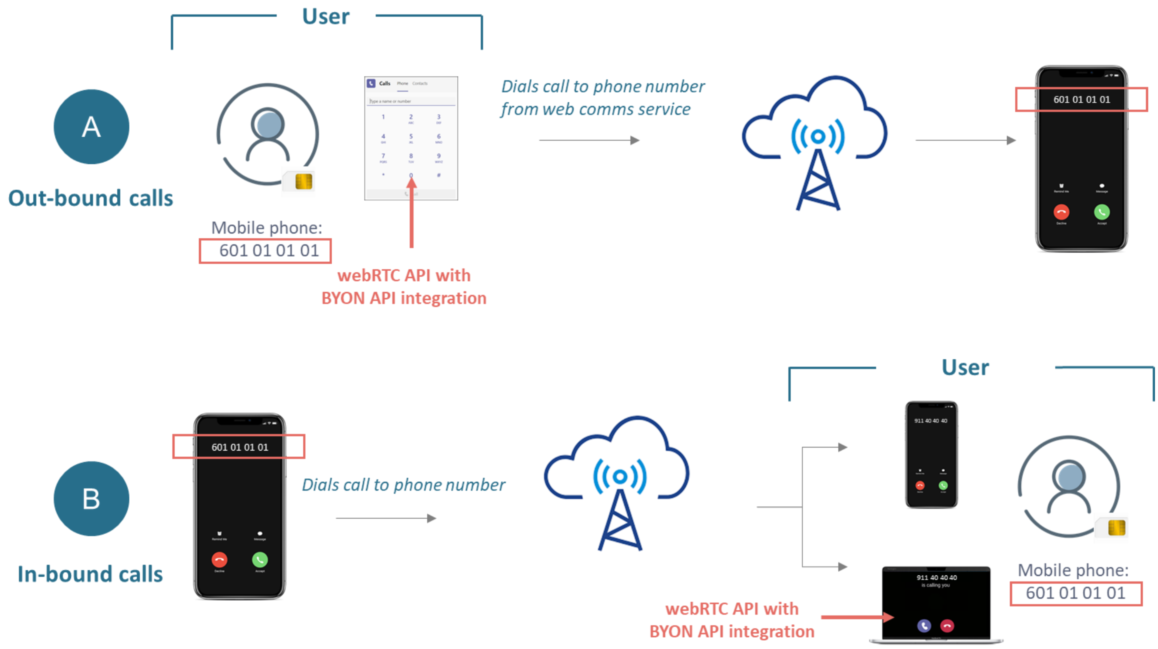

FIGURE 1: IN-BOUND AND OUT-BOUND CALLS FLOW

For an average user the main benefits are related with the simplicity of creating a communication environment involving different access points with SIM-less and no-SIM-less devices just with having a single MSISDN.

4.Use cases / user stories

Down to earth, the use cases that come along with this functionality, as specified above, are based on offering calling services with just a single or even no SIM card.

For example, in B2C use cases so users can connect to businesses customer care, sales point etc. In this case the called number is normally fixed by Business and the calling number is picked up from a pool of numbers assigned to the Business for this service. In that sense, the call can be done from any device that has a web browser environment, including laptops, PCs, XR immersive glasses, etc. This is an example of a non-SIM card WebRTC use case.

Click to call

For example, in B2C use cases users can connect to businesses customer care, sales point etc. In this case the called number is normally set by the Business and the calling number is picked up from a pool of numbers assigned to the Business for this service. In that sense, the call can be done from any device that has a web browser environment, including laptops, PCs, XR immersive glasses, etc. This is an example of a non-SIM card WebRTC use case.

Metaverse calls

Now describing a single SIM card use case. When enjoying an immersive experience, or engrossed in your favourite game, things such as important calls can go unnoticed. In order to interact with outside stimuli (i.e., take that call), the user must remove their VR headset, interrupting the experience, and then put it back on if they want to continue afterwards. This is less than ideal for something as immersive as a VR experience, where a frictionless engagement is crucial.

Metaverse Calls addresses the problem by enabling traditional voice and video calls in the Metaverse. We want to bridge the gap between both worlds by easing communication among them.

This use case brings the user’s phone number to the Oculus headset. With Metaverse Calls there’s no need to take the VR headset off every time the user hears the sound of a new notification coming in and can be aware of everything that happens in their phone without abruptly interrupting their immersive experience.

5.BYON: Functionality description

Considering the level of feature complexity, we could divide the functionality in different levels of complexity:

BYON Basic:

- Register an MSISDN and make and receive calls with the MSISDN forking (parallel ringing) to the SIM card-based smartphone and the registered web application. This is the functionality proposed in this document.

BYON Advanced:

- Hot Transfer calls from the web app to the SIM based smartphones and vice versa
- Provide the voice call status for the web app to change her presence status information.

BYON Complex features:

- Make and receive calls from the web app to the web app using the web account identity and *forking* them to do parallel ringing on the smartphone.

This first effort addressed by TIP MRN focuses on **BYON Basic features**. Therefore, the subsequent technical realisation and architecture description in this document describes this functionality.

6. Technical realisation

It is proposed to leverage on the IMS domain as future proof technology for Voice in telecom networks. The legacy voice technology based in circuit switched is not suitable for Metaverse calling

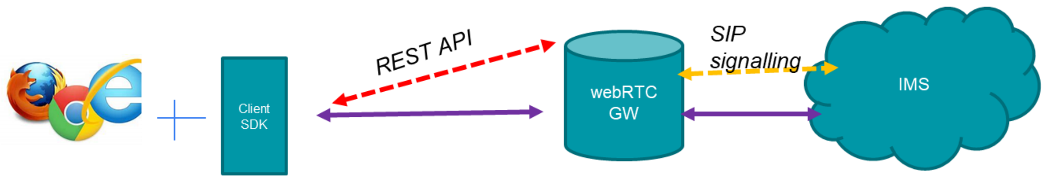

FIGURE 2: WEBRTC HIGH LEVEL INTERFACES

To accomplish this, the functionality that is already used for making voice calls is used, by leveraging the IMS Core that is already deployed. Clients on the user devices implemented the WebRTC protocol on the web browsers or can easily implement it.

BYON is initially addressing mobile voice services, and therefore, the focus is on the VoLTE service.

WebRTC Gateway (GW) enables the interworking between the web-based world and the IMS world, providing an API for the Control Plane of IMS specifically designed for web applications. Additionally, it offers a standard

WebRTC interface for media, allowing voice and video call services to be integrated into new apps through open gateway exposure. Therefore, the WebRTC gateway is the element that will be the bridge that connects the clients to the voice network. Its function is to interwork API, web protocols and Codecs to the protocols used in the network.

The WebRTC GW facilitates UNI (User Network Interface) access to the network by providing a single number for VoLTE services and enables NNI (Network to Network Interface) access to the network through click-to-call functionality. And these functionalities can bring new use cases to the Metaverse.

For providing authentication functions to web clients, an IdP in the operator network will be used. IdP (Identity Provider) is the Identity Provider, it will be in charge to authenticate, authorise and give access to the client to the IMS Core.

The IMS Core will need to offer multi-device capabilities. The standard way is to be able to register multiple identities and do forking. The protocol used for signalling are SIP/SDP and the media is RTP o SRTP.

In the negotiation of the call, the WebRTC GW and the IMS need to negotiate a Codec used for communication. Depending on the WebRTC GW, the transcoding must be done in the SBC or in the WebRTC GW. The Codecs used in WebRTC are not the same as the VoLTE service.

7.Architecture

7.1. E2E Architecture diagram

The elements involved in the integration of the WebRTC GW are:

- **SBC**: as the border element of the IMS voice domain. Normally the webRTC Gateway will integrate to an Access SBC but also it is possible that this function is incorporated within the WebRTC Gateway. It depends on the WebRTC vendors. Some vendors offer the WebRTC inside their SBC products and others as separate elements. Some others do not offer P-CSCF function on their WebRTC, so it needs to integrate to an SBC of the IMS Core. The SBC must accept the use of SRTP. This means basically there are two integration options (Mw with SBC and Gm with CSCF). Operators will normally choose to reuse the existing SBC and therefore access the security layer and opt for the Mw based integration option. When deploying this Mw option, transcoding needs to be observed. It is expected that the webRTC has transcoding capabilities.
- **IMS CORE:** It needs to be configured to use and store multiple SIP REGISTRATION from different devices. Also needs to support Forking of SIP signalling or an AS for multi-device.

- **IDP:** clients using the WebRTC in most cases are SIM-less, so traditional IMS Core credentials cannot be used. In this case, we will use an Operator IdP that is able to authorise a client to access the IMS service. The idea is to use Web Based authentication.
- **Clients:** Each vendor has their own implementation for the communication to a client. Some use REST API, and others implement an SDK that needs to be installed in the device. An SDK that will encapsulate the vendors API and give one unique API that developers can use without knowing what WebRTC vendor is used is being developed. The idea is to bring that API to CAMARA.

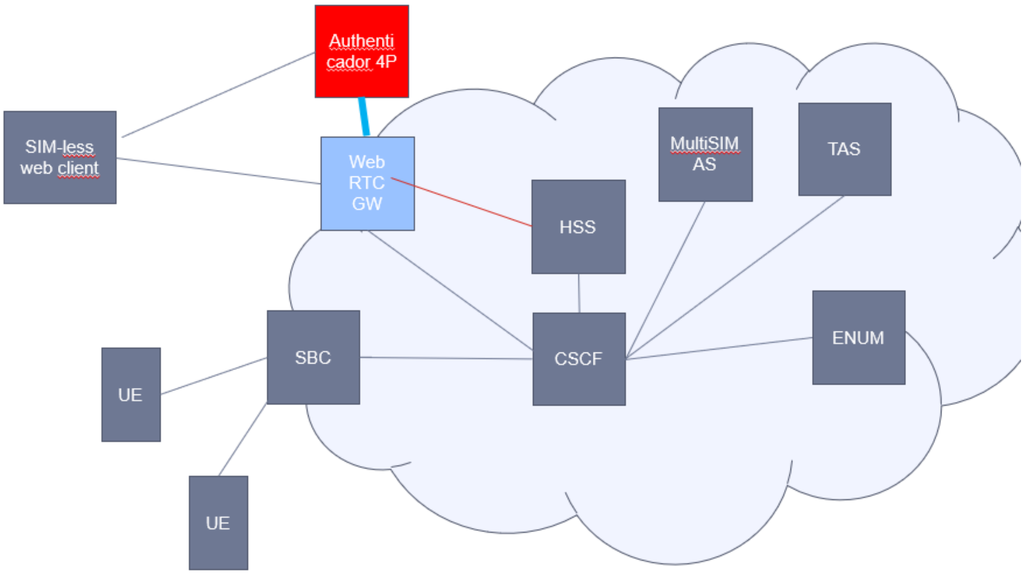

FIGURE 3: DETAILED ARCHITECTURE

7.2. Architecture details on webRTC components.

The control plane API or interface for the WebRTC GW can be implemented using JSON over WebSockets, SIP over WebSockets, or a RESTful JSON API. These options provide the necessary protocols for communication between the web-based applications and the GW's control plane.

For media transmission, the WebRTC protocol with SRTP (Secure Real-time Transport Protocol) is utilised. This ensures secure and reliable delivery of voice and video streams between the web-based clients and the GW.

Notifications can be delivered using WebSockets or through Push Notification systems designed for Android and iOS platforms. These mechanisms ensure that users receive timely notifications about incoming calls or other relevant events.

To facilitate integration and development, the vendor can provide an SDK (Software Development Kit) that includes necessary tools and resources for developers to utilise the GW's features. Additionally, a testing web client may be provided to assist in testing and validating the functionality of the GW.

To handle user authentication and identity management, the GW must have interfaces to an Identity Provider (IdP). This integration allows the GW to securely authenticate users and manage their digital identities during the registration and authentication process

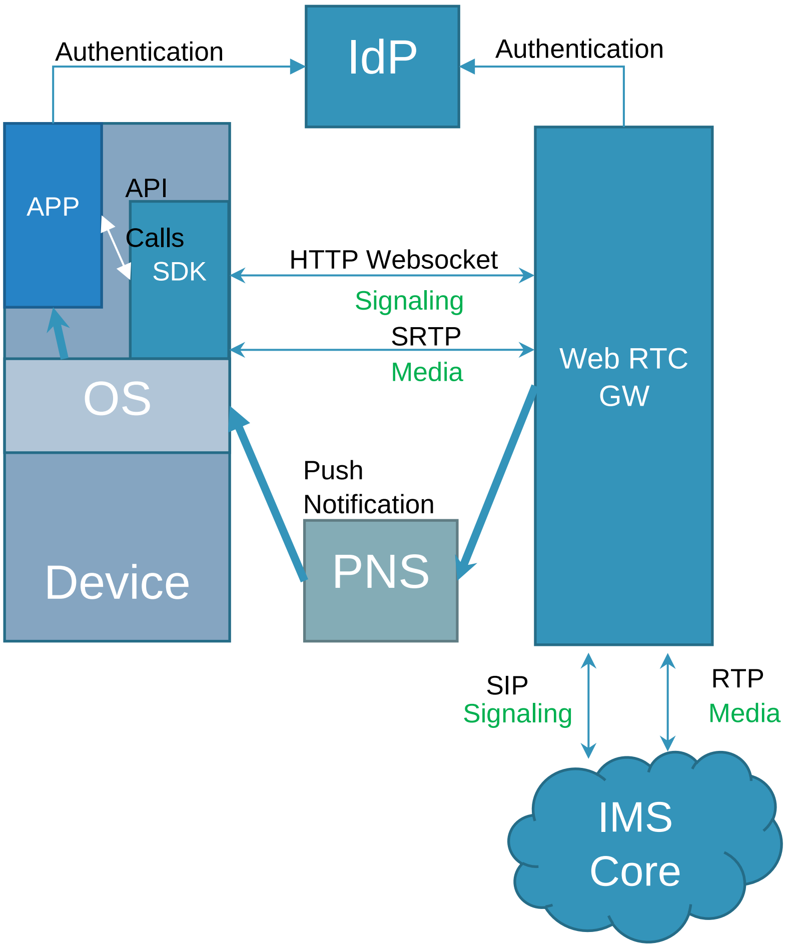

FIGURE 4: WEBRTC LOW LEVEL INTERFACES

7.3. Benefits of webRTC-based approach

There is a growing demand to bring Comms services beyond SIM based devices and support a variety of application clients that rely on http (web-based client). Bringing Telco Comms to immersive experiences and merging Telco Comms with web based (OTT) Comms.

Using a WebRTC GW approach provides the following benefits:

- Easiness and flexibility on implementation are one of the biggest advantages of using WebRTC over, for example, a straight SIP integration. WebRTC can be built into modern web browsers, eliminating the need for additional plugins or software installations.
- In the same way, webRTC seamlessly integrates with other web technologies, such as HTML, CSS, and JavaScript. This allows developers to create rich, interactive web applications that incorporate real-time

communication features. Direct SIP integration, on the other hand, may require separate software or clients to enable communication, which can add complexity and limitations.

- Simpler integration relying on the A-SBC of the network, does not require complex or deep integration inside the core or modifying the HSS user profiles or VAS services provided on top of VoLTE (unaffected orchestration logic).
  - Routing and forking functionality are handled natively by IMS
  - At Operative level, security and stability operation policies in the operators normally prevent direct integration to critical elements such as CSCF/IMS core, and mandate integrations based on border elements such as SBC. Therefore webRTC-GW-based integration complies with this.
  - This provides a faster route to technical readiness and TTM.
  - Web applications can integrate using REST APIs for Identity management and registration and call handling.
  - It is possible to provide a single-entry point to operators with multiple operating businesses.

7.4. Functionality required in existing network elements.

To offer this functionality, there are some features that are needed in the IMS Core.

The IMS Core needs to be able to store more than one Registration information for the user. The IMS Core will receive from the primary device and from the WebRTC client SIP Register with the same IMS Identity, but with different IPs and SIP instances.

In addition, with the above, the clients need to be able to fork IMS signalling to each of the registered devices. The CSCF must have this feature activated to be able to fork de invites to each device and control the dialog of the forked signalling.

Depending on the implementation of the WebRTC GW, the SBC will need to do the transcoding of WebRTC Codecs to VoLTE codecs. But it could be that the WebRTC GW already does this transcoding, especially if reusing existing SBCs that might not be planned for this transcoding in terms of capacity/dimensioning or capabilities.

Depending on how the IdP identity is implemented, it may be needed to have the WebRTC GW as a trusted node in the P-CSCF and S-CSCF to avoid authentication challenges.

7.5. Requirements on devices/clients

The clients need a web browser or an application with WebRTC library, also it will need to have Voice capabilities.

Depending on the implementations of the WebRTC GW it could need an SDK to connect to the WebRTC GW.

Please notice that webRTC libraries in the client use free codecs like OPUS, while the voice network uses AMR. Thus, transcoding is expected in the web RTC interworking architecture in the network.

8. High level flows reference

8.1. User authentication and webRTC gateway discovery

The user has a subscription to an ISP/operator. They use their mobile phone number to make and receive calls from her device, but as well to any non-SIM card device (web browsers)

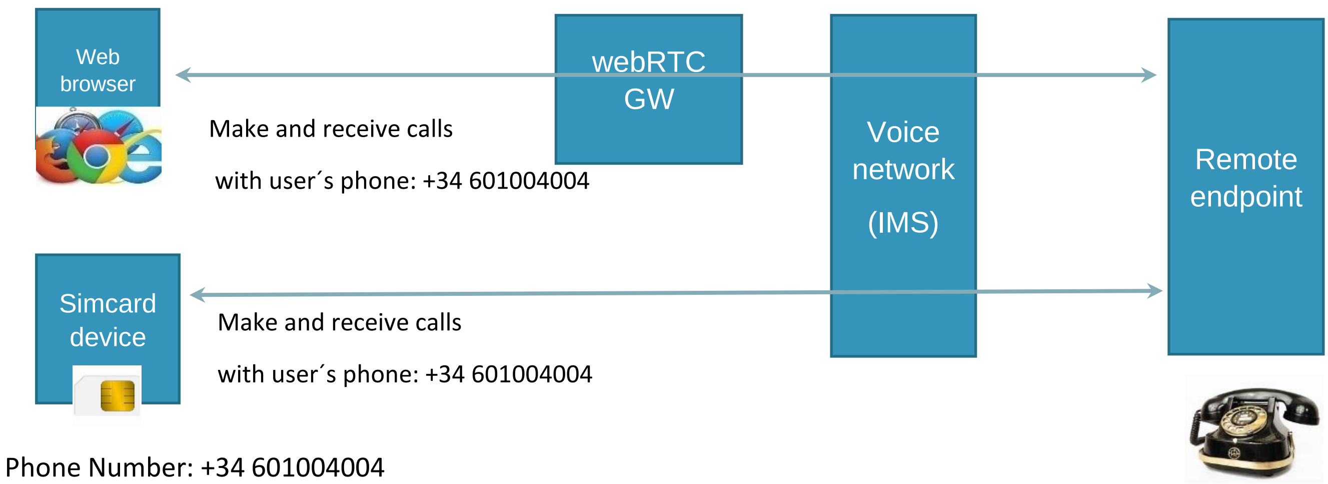

FIGURE 5: USER AUTHENTICATION AND GATEWAY DISCOVERY HIGH LEVEL FLOW

8.2. Registration

A Web Browser app using the WebRTC BYON SDK wants to register from this device. The GW will request credentials, and through the SDK the app will contact the ISP/Operator IdP and the user will authenticate using credentials that the user uses in other ISP/Operator Digital services. Besides authenticating the IdP will provide the user phone number that the WebRTC GW needs to use to register the user in the voice network.

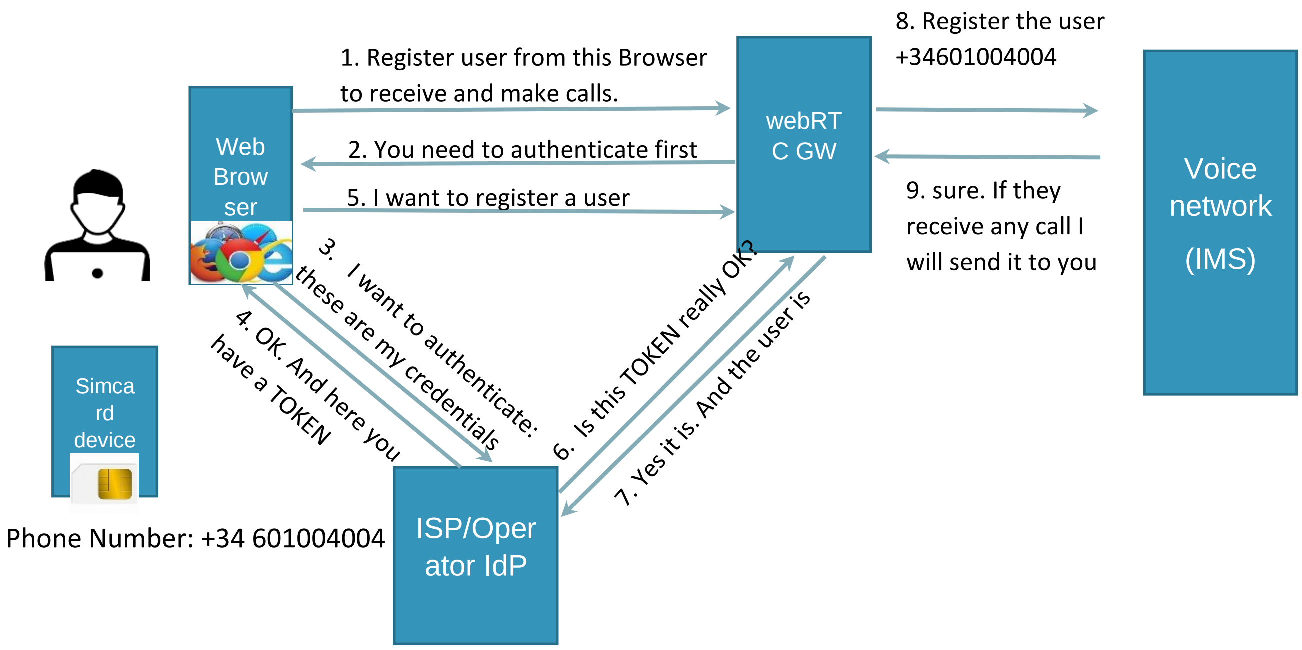

Figure 6: Registration high level flow

8.3. Make calls

The user can make calls using their number from their SIM card device or from a browser-based app like in this example.

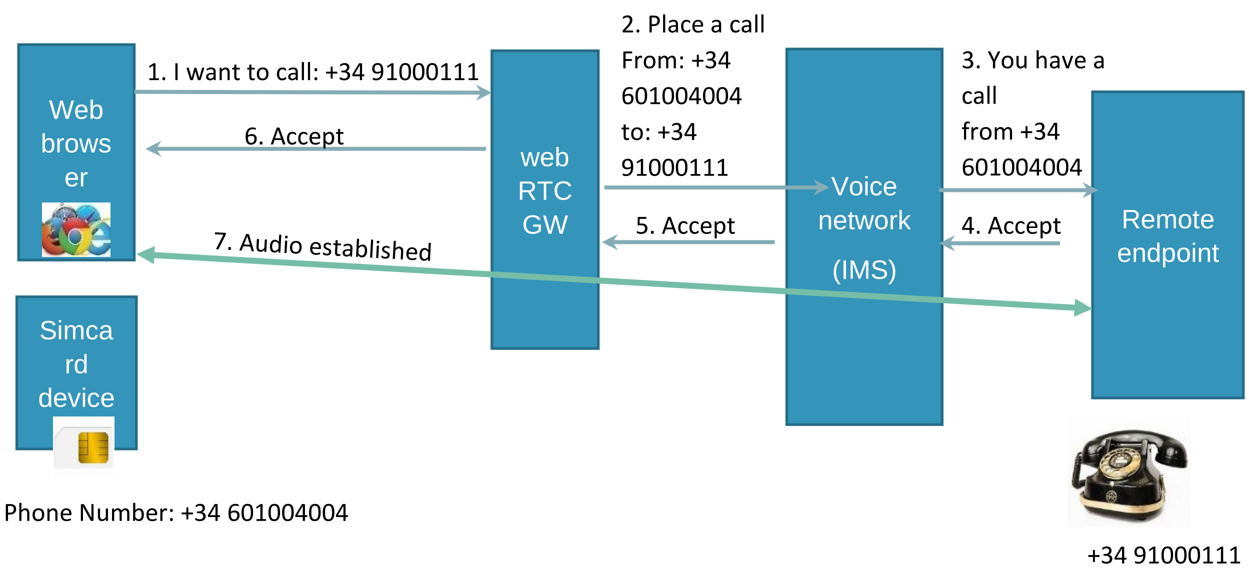

Figure 7: Make calls high level flows

8.4. Receive calls

When receiving calls, the Voice Network will see that the user is registered from a web application and from a SIM card device and will fork the call to the SIM card device and to the webRTC GW that registered the user. There will be a parallel ringing and the user can choose which device to use. In this example she chooses the web-browser app.

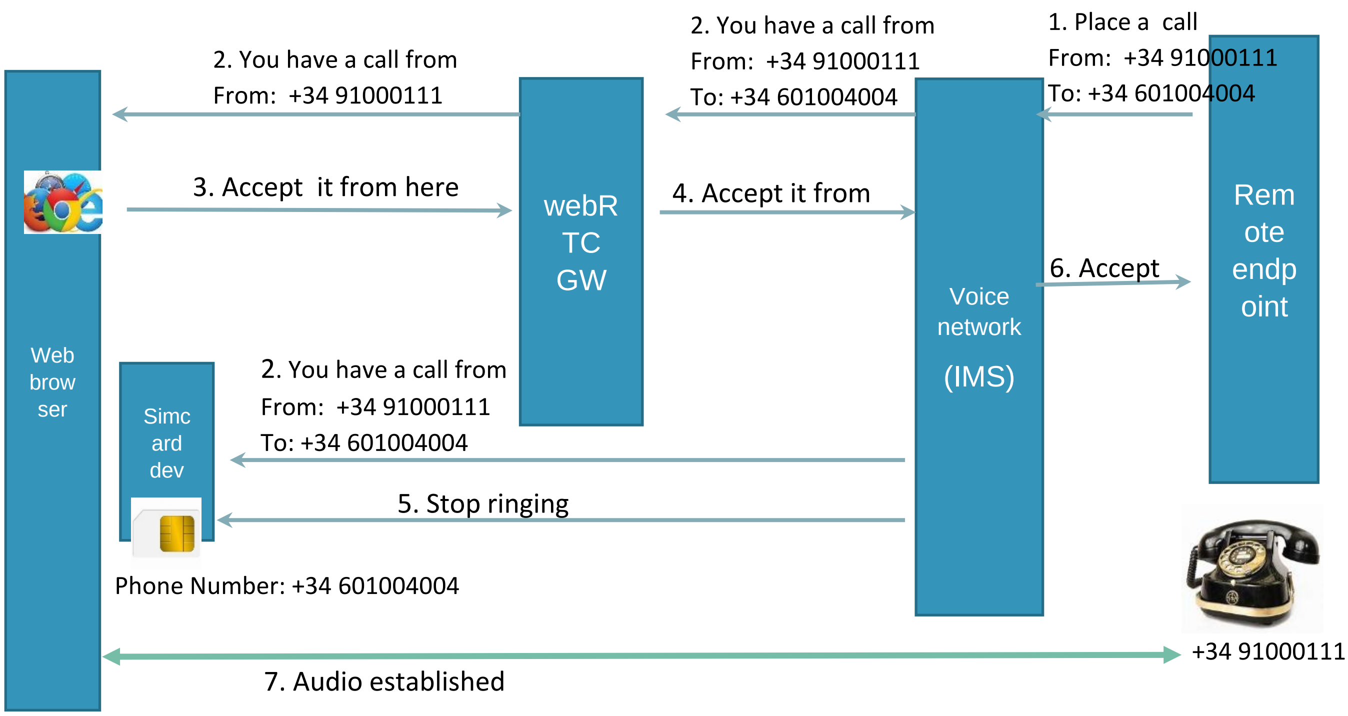

FIGURE 8: RECEIVE CALLS HIGH LEVEL FLOW

9.Low level flows reference

9.1. User authentication and Registration

The user authentication will be done using OAuth 2.0 using Operator provided Identity Provider (IDP). Once the client has been authenticated using the user provided credentials, the client will register with the IMS network, which is a pre-requisite for any IMS network access. IMS Network will also require periodic registration which will be done by the WebRTC GW without any need for a periodic trigger from the client. However, the client will need to periodically refresh the Access Token retrieved as part of the authentication process.

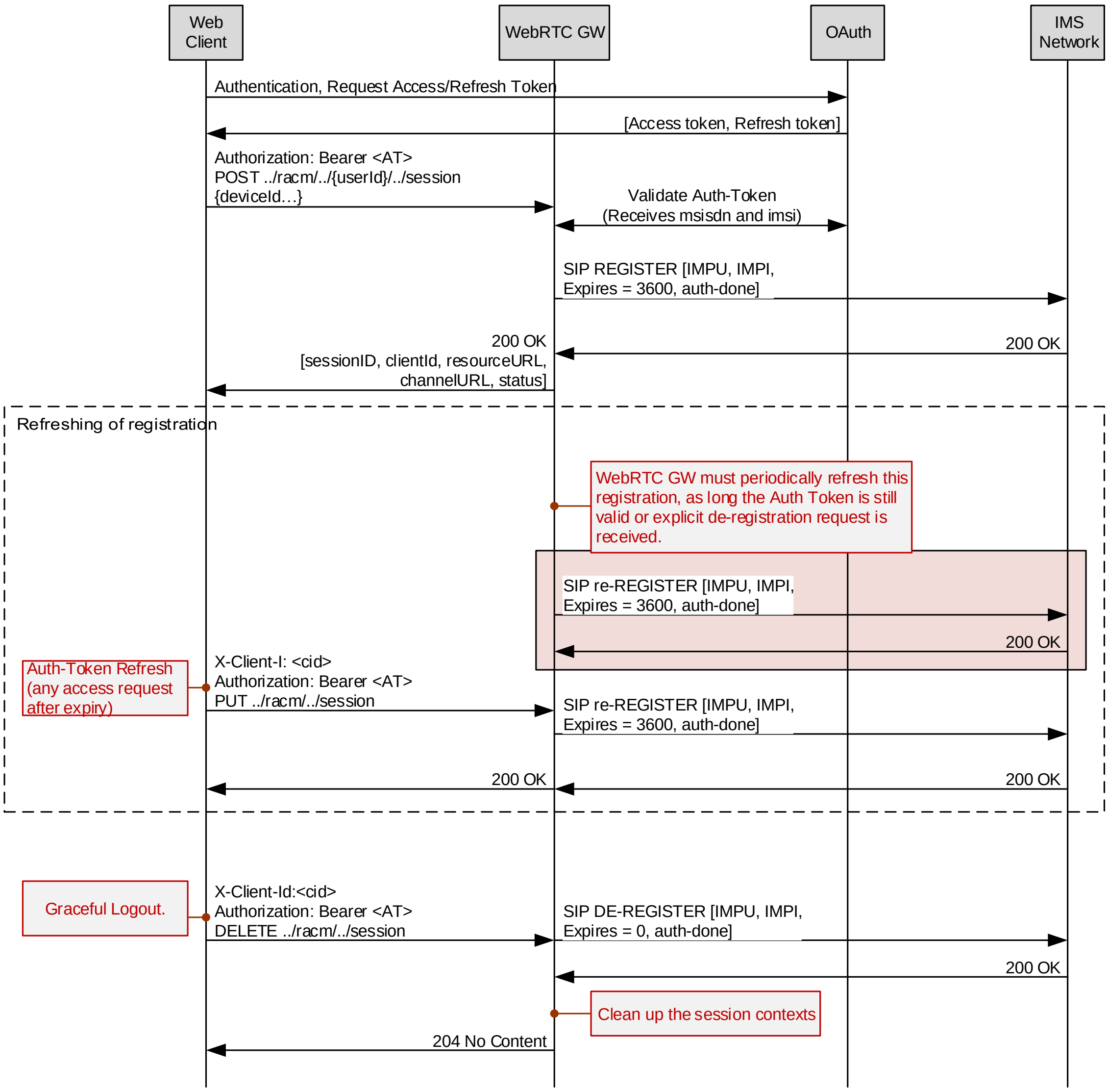

FIGURE 9: AUTHENTICATION AND REGISTRATION LOW LEVEL FLOW

9.2. Call Origination

Following a successful authentication and registration into the IMS Network, the user is ready to make an outgoing call. The client will establish WebSocket (WS) using the channel URL received as part of the registration procedure. This WebSocket connection will be used to receive notifications from the network indicating the progress of the outgoing call.

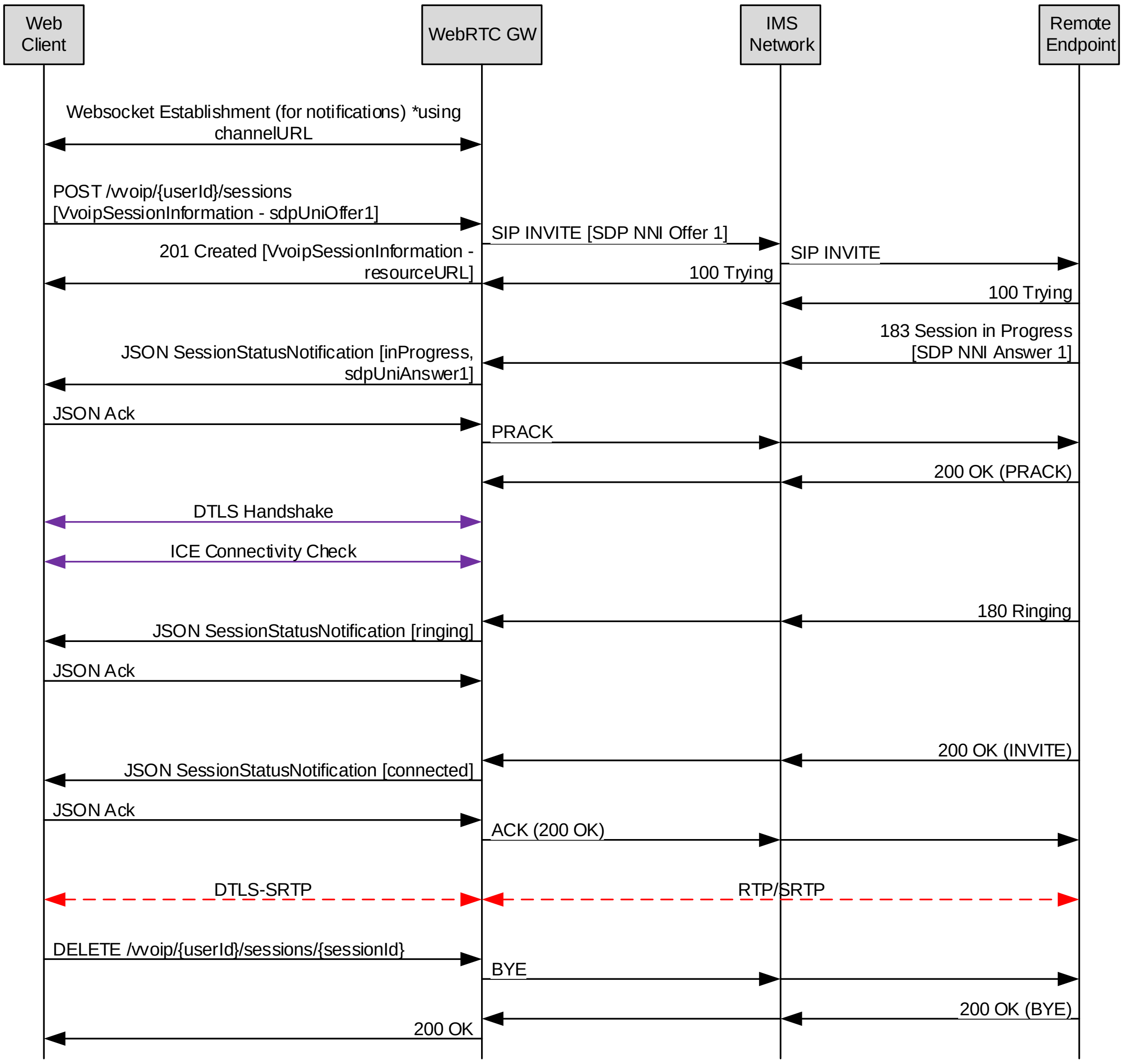

FIGURE 10: CALL ORIGINATION LOW LEVEL FLOW

9.3. Call Termination

The pre-requisite for call termination is the WebSocket connection establishment, as that channel will be required to notify the web client of an incoming call. This WebSocket connection ideally must be established during the initial registration process, to receive any incoming calls. The WebRTC GW will create the call session object prior to the notifying the end client of the incoming call. The web client will then modify this session object's call status to indicate the status of the ongoing call.

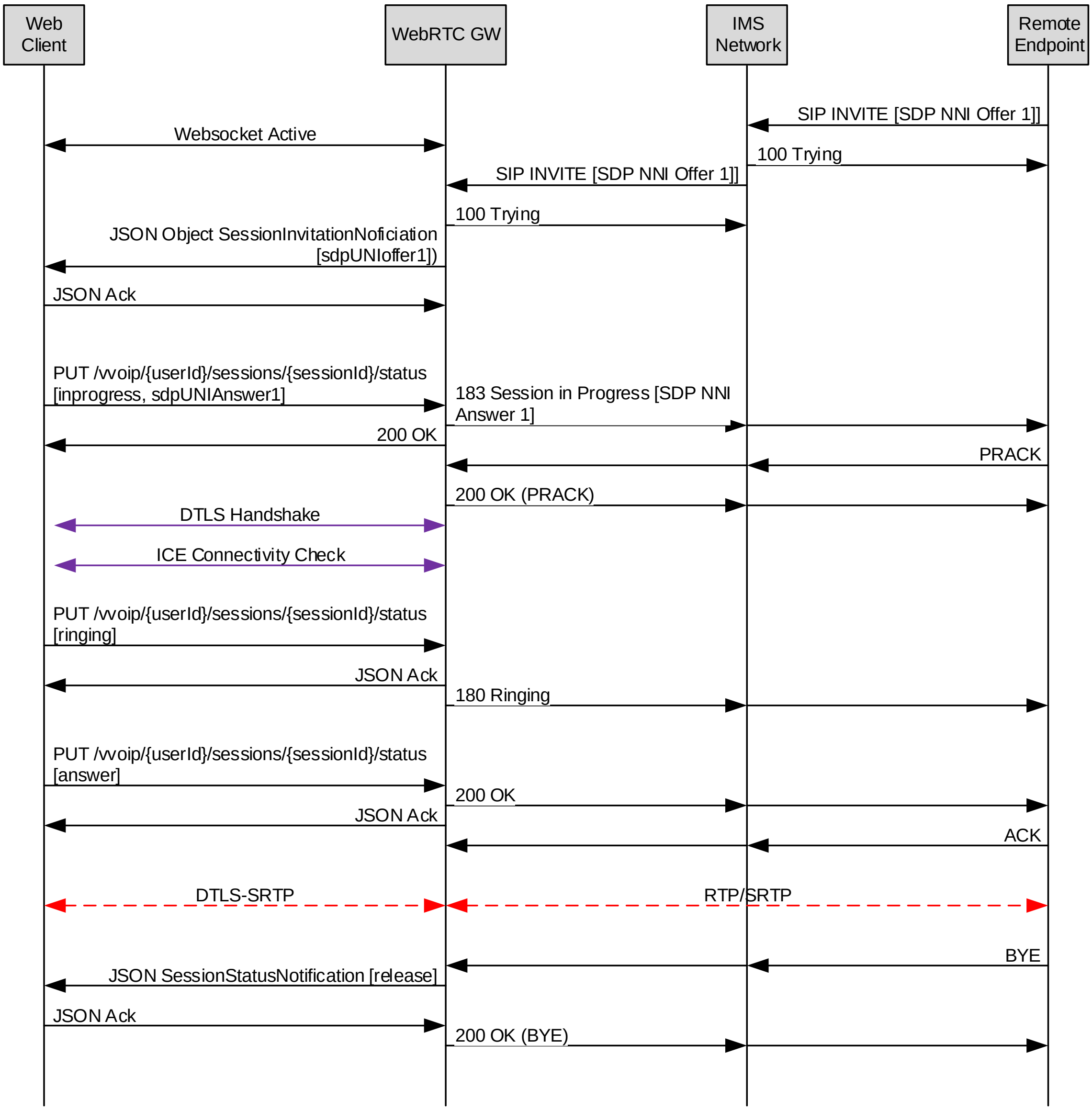

FIGURE 11: CALL TERMINATION LOW LEVEL FLOW

10. Standardisation status

Reference to the main existing standards for the technical pieces of BYON e.g., 3FPP TS, W3C specifications

This service is positioned within valid releases of 3GPP (3rd Generation Partnership Project), GSMA (Global System for Mobile Communications Association), OMA (Open Mobile Alliance) and RFC (Request for Comments) standards. Particularly related in the following standards documents:

- IR.92: “IMS Profile for Voice and SMS”
- IR.94: “IMS Profile for Conversational Video Service”
- RFC 2976: "The SIP INFO method"
- RFC 3261: “SIP: Session Initiation Protocol”

- RFC 3262: "Reliability of provisional responses in Session Initiation Protocol (SIP)"
- RFC 3265: "Session Initiation Protocol (SIP) Specific Event Notification"
- RFC 3311: "The Session Initiation Protocol (SIP) UPDATE method"
- RFC 3428: “Session Initiation Protocol (SIP) Extension for Instant Messaging”
- RFC 3903: "Session Initiation Protocol (SIP) Extension for Event State Publication"
- TS 23.380: IMS Restoration Procedures
- TS 23.228: IP Multimedia Subsystem (IMS)
- NG.134: IMS Data Channel

11. Gaps vs Standards

The final scope beyond this document is to bring this proposal to the industrial groups where the standardisation of this Control interface for webRTC-to-telco communications may happen.

The UNI interface for the control plane needs to be defined and standardised. It’s important because this interface manages registration and is in charge of making and receiving calls.

The Vendor proprietary APIs today are normally based on technologies as REST APIs, JSON over WebSocket or SIP over WebSocket. Making each solution work in a different way.

BYON API is basically a simplification of the existing webRTC control plane API from OMA that intends to reduce complexity for non telco developers who want to enjoy this service.

This API proposal has been presented and fortunately accepted in CAMARA group. The technology that has been chosen for this API is REST, utilizing JSON over WebSockets for the delivery of asynchronous call events.

Once standardized, numerous use cases can be carried out.

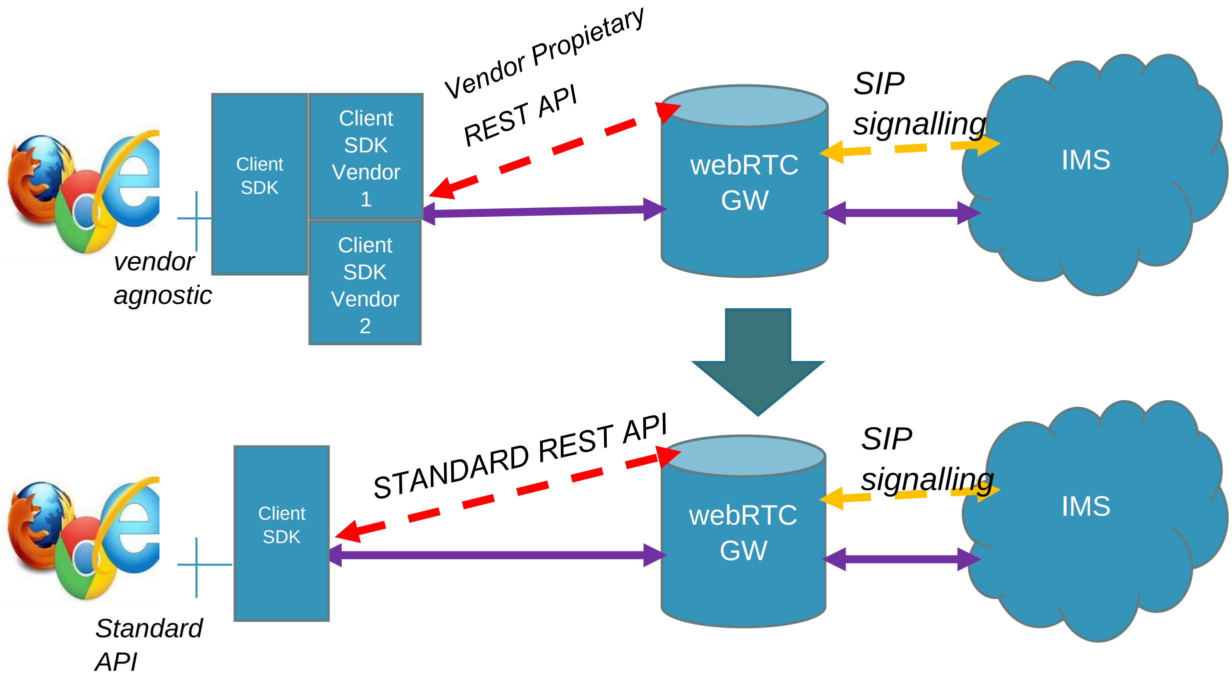

FIGURE 12: STANDARDISATION PROCEDURE

Disclaimers

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