SIPWISE AND DEUTSCHE TELEKOM / TLABS

ANNOUNCE COOPERATION FOR THE

WebRTC SOLUTION

RTC:engine

Sipwise and Deutsche Telekom AG / Telekom Innovation Laboratories signed a cooperation agreement for joint development and distribution of the WebRTC based communication solution RTC:engine. Sipwise includes the solution in their product portfolio and will provide vertical products and solutions (like call center, e-learning, …) on top.

Vienna, July 1st 2015

Executive Overview

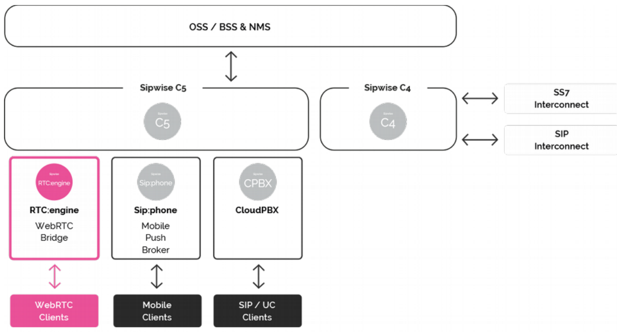
During the last year, Sipwise and Deutsche Telekom have been working together in the emerging field of web based real-time communication, enabling end users to consume voice, video and chat services via cloud based solutions. It is our common goal to completely uncouple a user's identity from their devices, granting access to services from wherever a web browser is available.

Sipwise as a vendor of Telco solutions has followed a revolutionizing approach in the industry since their market entry in 2008 by bringing the open and modular product development approach of the Internet world to the Telco space. Using and creating open source software to build cutting edge solutions for real-time communication has been key to rapidly launching stable and flexible products for Telco carriers who provide core- and OTT services to their end customers.

Since July 2015, Sipwise offers a new real-time communication framework called RTC:engine, jointly developed by Sipwise and T-Labs, the Innovation Laboratories of Deutsche Telekom. RTC:engine provides both back-end technology for high performance signaling and media routing, and web services and CDKs, allowing easy front-end application development in HTML5 for rapidly building cutting edge cloud communication applications.

A complete software-only approach encapsulating different signaling protocols and media handling in simple APIs with a focus on on-premise scenarios allows operators of RTC:engine to launch next generation communication services within their own private cloud infrastructure.

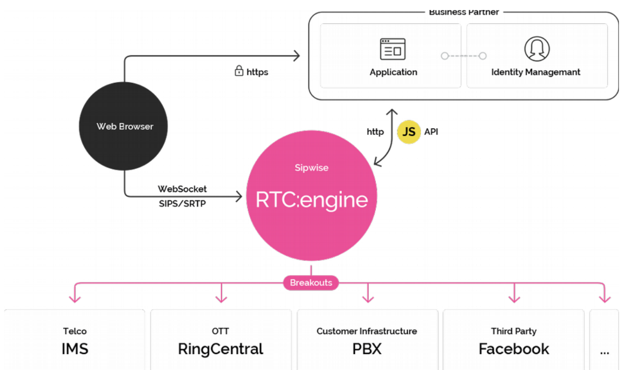
RTC:engine seamlessly blends into the Sipwise product portfolio and enables a new approach to communication solutions.



The key USP of RTC:engine is its capability to extend traditional communication infrastructure like SIP and XMPP services and enrich it with purely web-based scenarios in a very short time-to-market, without the need to dive into low level APIs or protocols.

Enabling new Use Cases

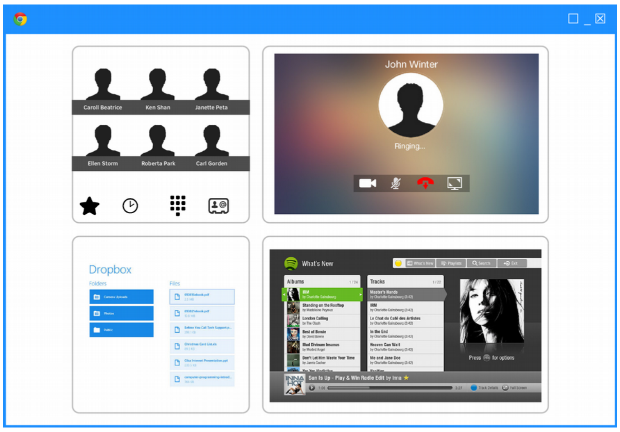
RTC:engine as a purely web based overlay on top of existing communication infrastructure enables new use cases for both core and OTT services.



Use Case: Cloud Telco

A purely web-based communication solution offering voice, video and instant messaging from within an HTML5 web application fully decouples the service from the device. No software or app other than a modern web browser needs to be installed on a device, nor is any specific hardware phone necessary to use the service. Yet, all typical features like call forwarding, hunting etc. are available as usual and existing end customer devices and apps can be used in parallel and seamlessly along with the web application.

Integration of the web app into existing customer portals with single-sign-on mechanisms allow for stronger customer loyalty. At the same time, third party offerings like cloud storage, music streaming and more can be used along with the real-time communication service, thus forming a fully featured entertainment and communication portal for end customers.



Use Case: Smart Video Sharing in Call Centers

RTC:engine offers the flexibility to establish two-way communication paths via arbitrary signaling channels, and therefore a side channel communication in addition to standard voice calls e.g. via mobile phones can be established at any time.

A compelling use case would be a call center agent creating an on-demand video call by clicking a button in a CRM console, which sends a link to the calling customer via SMS . Once the customer opens the link, a video element is shown in the agent's CRM, allowing him to instruct the customer to remotely show the cabling of a router or any other device the customer needs support for, and providing the ability to remotely take pictures which are automatically attached to the support case.

Use Case: Multi Party Video Conferencing and Collaboration

The classical use case for web based communication is cost effective multi-party video conferencing solutions without the need for tailor-made third party conferencing. Since video streams are sent separately to the web browser instead of having fixed tiles in one single stream, the web application can take advantage of designing the user interface according to their use cases, like annotating video streams and more.

Parallel instant messaging sessions in conferencing mode or in private chat sessions between participants, including peer to peer or public file sharing as well as screen sharing enrich the overall video conferencing experience.

RTC:engine and WebRTC

RTC:engine is based on WebRTC, an open source media engine for modern web browsers. WebRTC provides Javascript APIs for web applications to access the user's microphones and cameras on the one hand and to establish peer-to-peer connections between two browsers for exchanging real-time communication streams (voice and video calls) and data streams on the other hand. WebRTC focuses on security by enforcing strict encryption of communication channels, and on media quality by applying quality feedback and adaptive bandwidth control of codecs used.

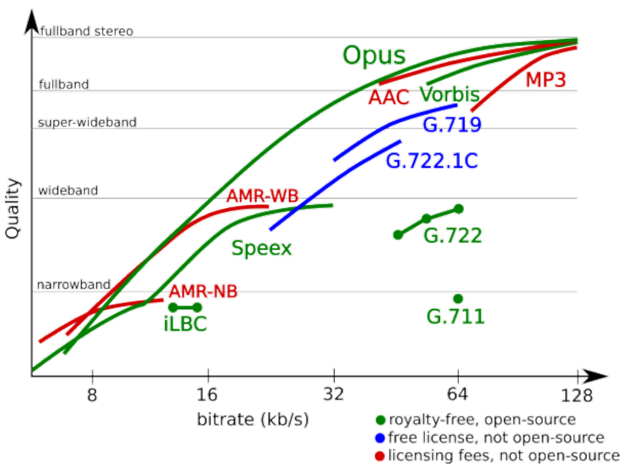
WebRTC in itself does not provide a signaling protocol to exchange media information between endpoints. This part is handled transparently for the application developer by RTC:engine by exposing a Javascript API in the RTC:engine CDK to perform the relevant communication tasks via HTML5 websockets.

The RTC:engine back-end system converts the internal signaling protocol to SIP, XMPP and other protocols which might be introduced in the future. This allows an RTC:engine web application to integrate seamlessly into existing SIP and XMPP networks. The encrypted media streams on the WebRTC layer are transparently converted to plain media streams by RTC:engine for communicating with legacy devices not supporting media encryption.

In case of internal communication scenarios or calls from or to SIP devices supporting the Opus codec, the media quality is full HD, providing crystal clear voice quality.

RTC:engine in a Mobile Environment

Due to the underlying WebRTC technology leveraging Opus as audio codec, the required bandwidth for calls using RTC:engine can be reduced significantly down to narrow band, resulting in bandwith consumption lower than AMR-NB and matching the bandwith of G.729. Transcoding between narrow-band on the RTC:engine side and G.711 for example on the peering side towards fixed and mobile networks is handled directly within the system.



This transcoding mechanism allows for clear voice calls even in scenarios with restricted bandwith like 3G and 2G, whereas full HD quality can be delivered over LTE.

RTC:engine and Native Apps

Implementing a soft-phone with voice, video, chat and presence functionality in general, and a mobile app in particular, require huge development effort due to their complexity and platform fragmentation (e.g. Android vs. iOS). This results in huge costs and slow time-to-market. Re-branding an existing solution requires a Telco or Enterprise to stick to the general outline and feature set of an App and only allows simple branding.

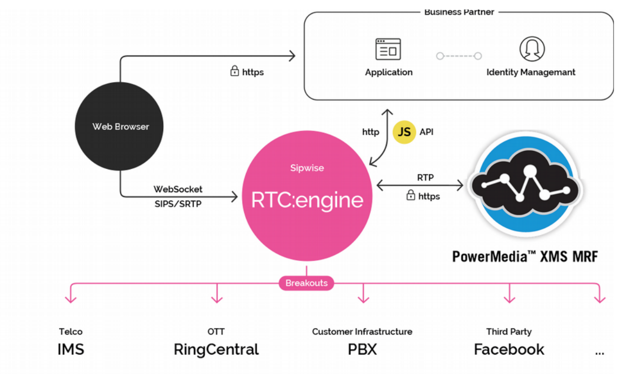
Alternatively, using a web based approach on top of RTC:engine gives full flexibility on the look and feel and the feature set included within the app, while at the same time it can be built to work on any platform (desktop browsers and mobile platforms). Releasing design changes and additional functionalities is as easy as rolling out a new web application.

RTC:engine based applications for mobile platforms are usually still wrapped within a very small native app in order to receive inbound call and message notifications when the customer is not logged in on the particular web page providing the cloud communication service. Sipwise provides mobile push mechanisms in the back-end to notify mobile devices, and a wrapper app which starts the mobile web application automatically or on demand to answer calls or replying to messages. On iOS, third party APIs can be linked into the wrapper app to expose the WebRTC APIs required by RTC:engine, which are currently not available natively on iOS.

Third Party Components - Mix and Match

RTC:engine is designed in a modular way to allow the engagement of different networks (native WebRTC, SIP and XMPP are provided out of the box) as well as different third party application servers.

One application server natively integrated into RTC:engine is Dialogic XMS, providing multi party video conferencing services, transcoding between various codec pairs for improved interoperability, and announcements.



Any combination of networks and external application servers is possible. New plug-ins are provided on request.

Contact Information

For more information on RTC:engine or the Sipwise product portfolio, please contact our Sales team at <sales@sipwise.com>.