

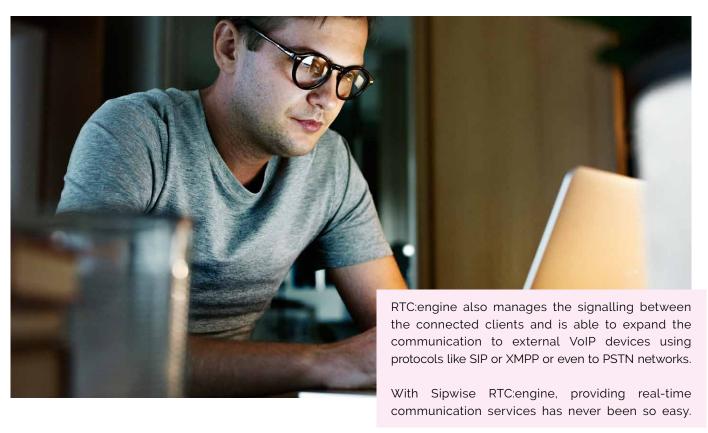


for Cloud-Based Real-Time Communication

# **Enabling Real-Time Communication Services with RTC:engine**

RTC:enigine is a WebRTC-based communication API. It simplifies real-time communication systems and enables rapid application development based on the RTC:engine interface. This allows you, as operators and/or over-the-top service providers, to offer instant real-time communication services in a web browser.

Your end customers can enjoy the benefits of having access to cloud communications, such as live video, voice, messaging and conferencing services on websites and mobile apps without installing any plug-ins or software.



## **Features**



#### Voice and Video

A simple set of API calls is sufficient to enable secure, peer-to-peer or anchored voice and video streams between web browsers without requiring the developer to know about the underlying signalling and media details. Both one-to-one and multi-party sessions can be established by means of a few lines of Javascript code.



Exposing the presence status of users via API call-backs enables customers to implement buddy lists into their web application, making it easy for users to determine the availability of their communication partners.

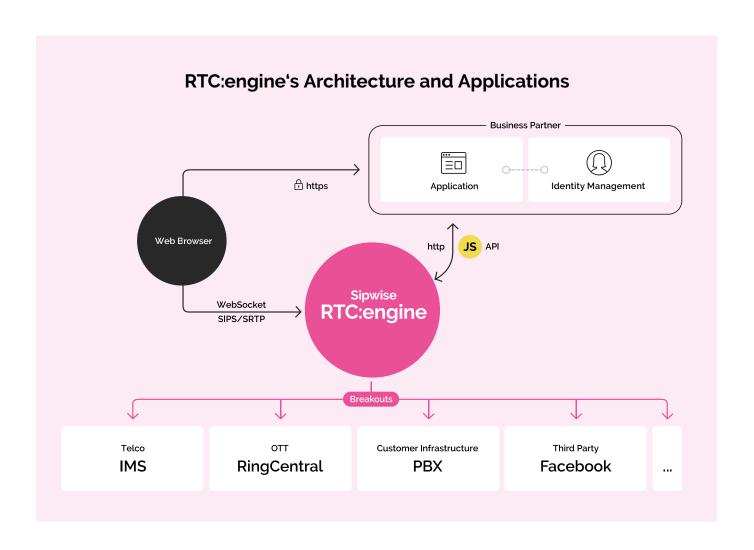


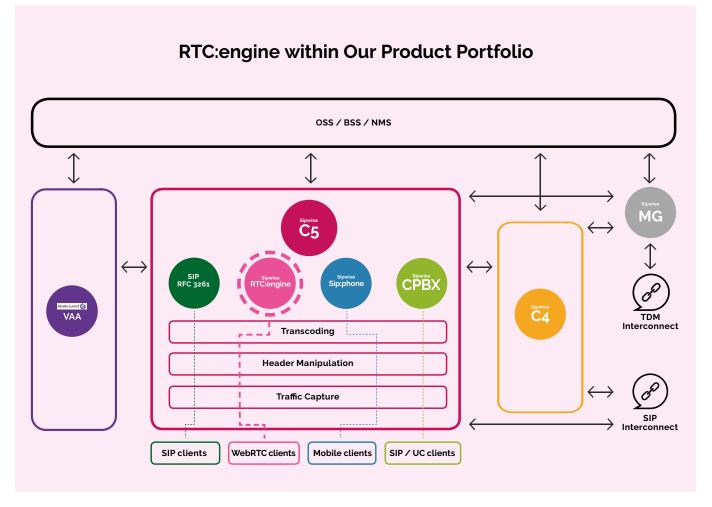
### Messaging

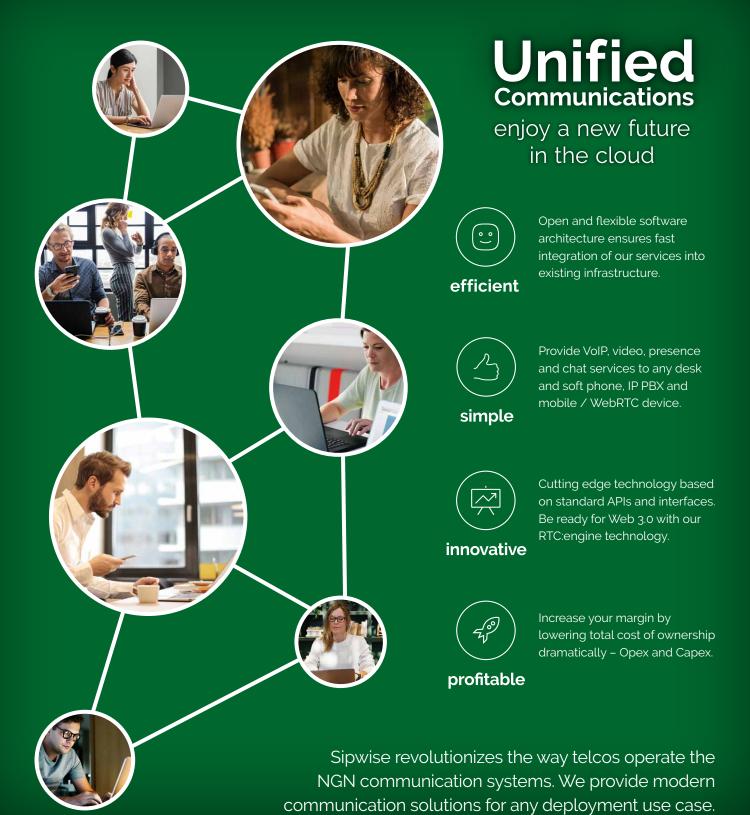
Exchange messages within chats and chat rooms in a protocol-agnostic way and allow to connect users over different networks using XMPP with GSM-SMS fall-back.



Media streams can be delivered via narrow-band or high-definition quality codecs, depending on the access networks. For example, WiFi enabled users can communicate in full HD, whereas participants on 3G with restricted bandwidth will fall back to narrow-band communication.









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# Need help? Contact us now:

Sipwise GmbH +43-1-252 152 3 sales@sipwise.com

sipwise.com

