Fact Sheet

Sipwise

RTC:engine

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PRODUCT SPECIFICATION

Product Positioning

RTC:engine is a WebRTC-based telecommunication API, allowing Operators and OTT Providers to deploy cloud services for end customers, like live video, voice, messaging and conferencing services on web sites and mobile apps.

Our lean and simple API takes away the complexity of real time communication technologies and enables rapid application development based on the RTC:engine interface.

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 signaling and media details. Both one-to-one and multi-party sessions can be established by means of a few lines of Javascript code.

Messaging

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

Presence

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

Transcoding

Media streams can be delivered via narrow-band, ISDN and 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 band will fall back to narrow-band communication.

Need help? Contact us!

sales@sipwise.com or give us a call +43 / 1 / 252 15 23 www.sipwise.com