WebRTC

Web Real-Time Communication

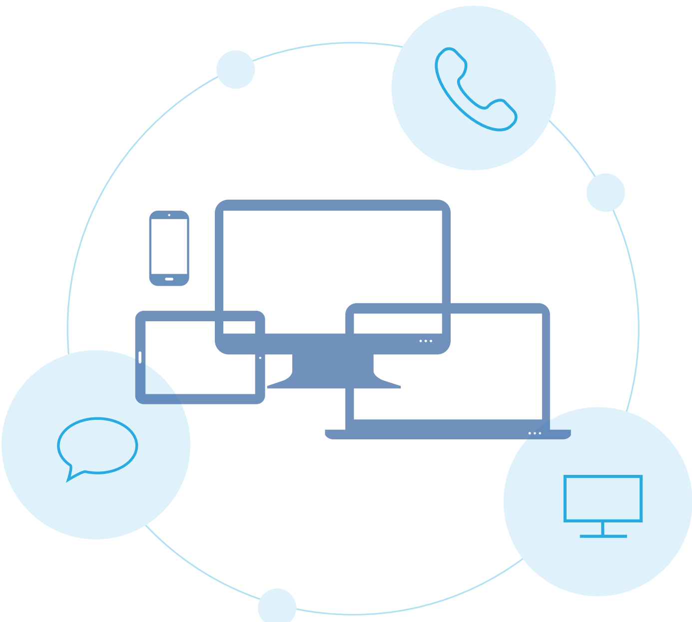
Call, Video Call and Chat

WebRTC is direct communication through the browser.

If your customer has access to a browser – Chrome, Firefox or Opera – he'll be able to communicate with the service center. By calling, video-calling or chatting – but above all encrypted and therefor absolutely safe.

A WebRTC can be directly embedded on a website and offers flexible choice between the communication channels voice, video and chat. It is therefore an easy and attractive plus that gives the user a very comfortable means of communicating with your enterprise.

It is suitable not only for internal corporate communication, but also for direct contact between the customer and a service center.



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COMMUNICATE WITH EVERYONE. EVERYWHERE.

General service specification

FUNCTION OVERVIEW

Multichannel communication
Audio- and video-calls and live-chat
Audio, video, chat conferencing

TEHNICAL SPECIFICATION

Supported platforms

-Backend: Linux
-Web App: WebRTC supporting browsers: Google Chrome, Mozilla Firefox, Opera
-Mobile Native App: Android, iOS*

TEHNICAL SPECIFICATION

WebRTC

-Web socket secure (WSS) connection

-Proprietary communication protocol between server and client

-Javascript Client

-NAT traversal when using STUN, ICE, TURN*

-Call API*

SIP

-SIP over UDP (RFC2833)

-SIP Back-to-back UA

-User registration

-Registration pass-through Modus

-DTMF SIP INFO

MEDIA SERVICES

G.711 A/Ulaw, G722, OPUS, H264, VP8 pass-through
Codec filtering
Dynamic jitter control
NAT/NAPT on media
RTP inactivity monitoring
Echo Test service

QUALITY OF SERVICE MONITORING

Dynamical bandwidth estimation and adoption

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HIGH AVAILABILITY AND SCALING

Active-active redundancy model
Distributed configuration
Dynamical scaling to fit load requirements

ROUTING

Partitioning (multi domains Support)
Call authorization
Routing by many parameters:

-URI: B-number+Domain
-A-number, source IP, transport protocol, source Domain

Call blocking and filtering
Embedded routing engine
External routing engine
Load balancing
Alternative routing on failure

MANAGEMENT

Secured Web-based UI for configuration and monitoring
Logging of alarms, events, statistics
Troubleshooting via UI
RESTful API
SSH access

SUPPORTED PROTOCOLS

WSS, RTP, SRTP, DTLS, RTCP, SIP UDP,
RFC 4585, RFC 3550, RFC 5104
Translation between transport protocols

HARDWARE AND VIRTUALIZING

Hardware independent
Runs on all virtualisation platforms