quobis: Janus: not just WebRTC!





About me

Graduated as Telecom Engineer

Working with Quobis and WebRTC since 2013

Backend developer

I'll never put pineapple in pizza

Quobis



50+ engineers working in UC



Integrating telco-class solutions since 2006



HQs in Galicia (Spain) worldwide operations



Working with WebRTC since 2013



Quobis Communication Platform



Industry awards & certifications

Quobis Communication Platform



WebRTC Application Controller (wac-core)



Unified Clients: Android, iOS, web and desktop



Signaling server (qss)



SDK: JS, Android and iOS



Janus WebRTC Gateway

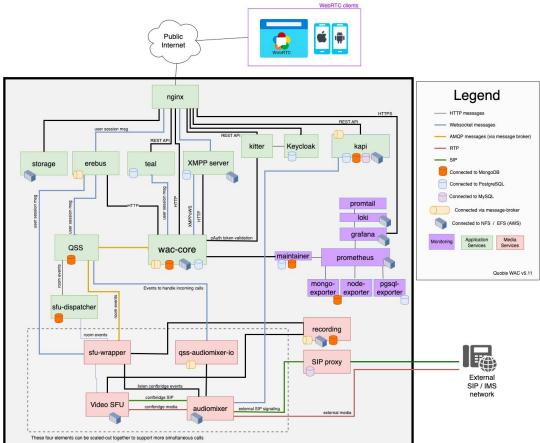


Other standard SIP components: Kamailio, Asterisk

QCP

An overview of our platform.



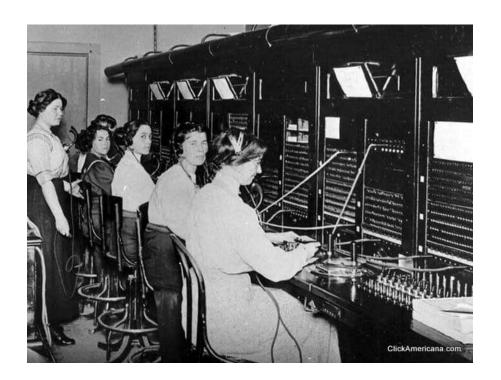


SIP trunk

SIP register

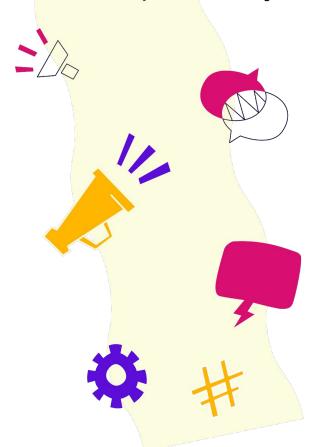
Parallel SIP





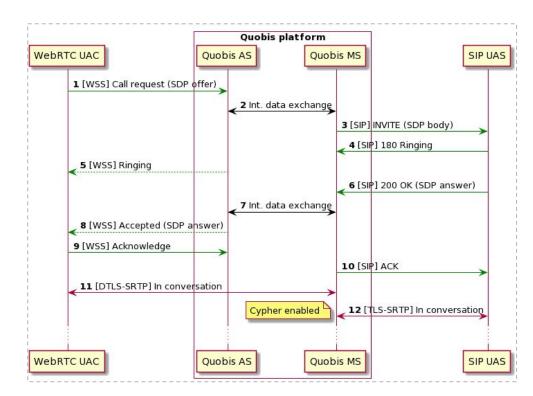
SIP Trunk

Trunk integration SIP header manipulation



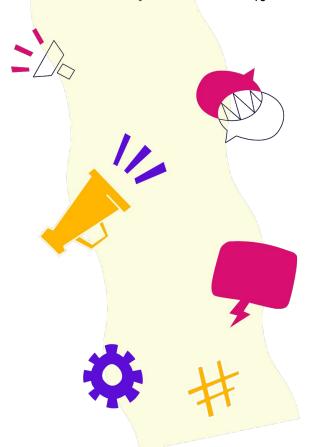
Call flows (WebRTC to SIP)

This image describes an outbound call from WebRTC to SIP.



SIP Register

SIP REGISTER
SIP header manipulation

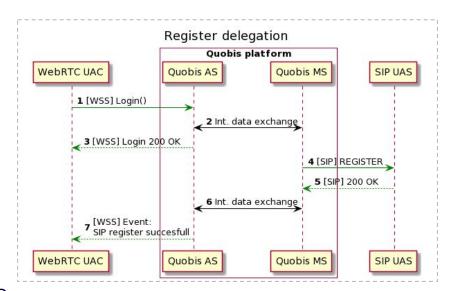


App login with SIP delegation (SIP REGISTER)

In case that customer requires REGISTER presence of the users, Quobis MS reflects app register as SIP register.

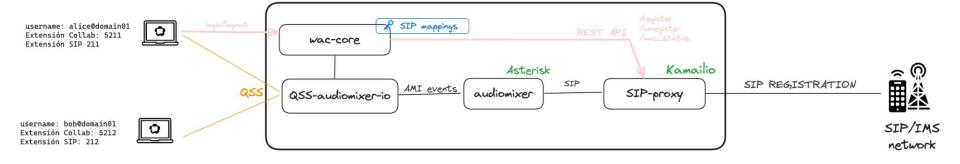
Customer can control SIP dialplan handling Quobis apps as SIP extensions.

On this basic integration, Quobis will keep internally WebRTC to WebRTC calls.



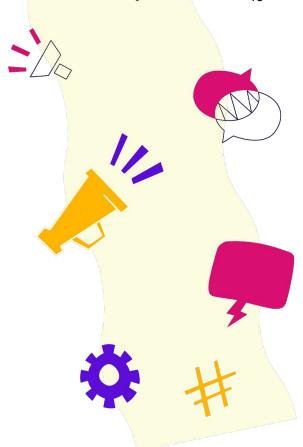
quobis:

Janus: not just WebRTC!



Parallel SIP

Full dialplan control on SIP side



quobis:

Janus: not just WebRTC!

14

Parallel SIP

If **parallel-sip function** is enabled, the calls between WebRTC users are sent via SIP to the customer PBX and, if the PBX decides to forward the call to Quobis because the callee WebRTC user is available, Quobis AS will detect it and will generate a video call between these participants.

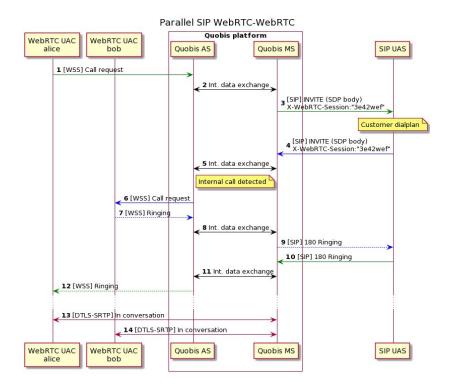
Benefits:

- Full SIP dialplan control
- Billing and control on a single point
- Signaling and traffic penalization

Parallel SIP

The outgoing INVITE request has a specific custom SIP header "X-WebRTC-Session" that contains an uuid, if it is returned in the incoming INVITE from SIP leg then Quobis can match the call from a WebRTC client and will add video using the Quobis MS.

Quobis MS will send the audio to the PBX, so it is possible to manage a conference room in the customer PBX side or in the Quobis MS side.



Demo?





Challenges

Three different modes of operation

Having flexible operating modes is always a source of pain when coding, testing and debugging

3rd party's infra

When using third party's infrastructure (ParallelSIP) there are increasing frictions

Numbering plan

Something not usual in pure WebRTC deployments

Grazie!



