

Sippo

Web Collaborator



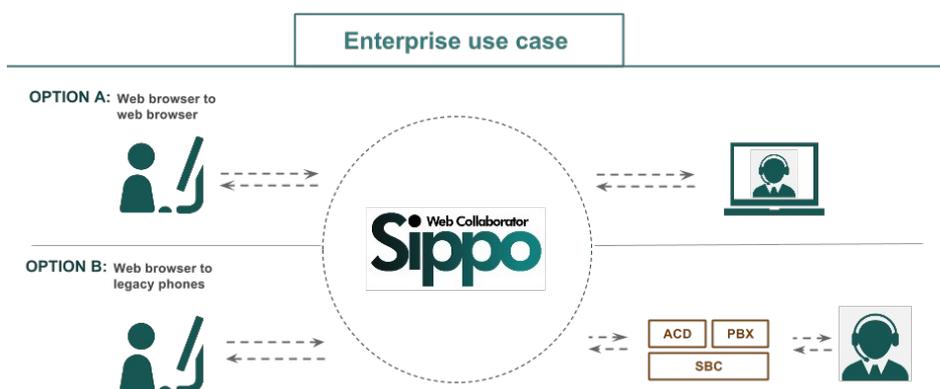
Overview

Sippo Web Collaborator is a fully-featured WebRTC-enabled corporate client with advanced capabilities that works with the Sippo WebRTC Application Controller (WAC). Built for the web with our telco background knowledge, Sippo allows IT managers of enterprises and service providers to deploy a lightweight web client without the hassle of installing any desktop application.

Sippo Web Collaborator provides a new way of collaboration, integrating the legacy audio features with new modern media-based communications: video, chat, file sharing, etc. In summary, a rich co-browsing experience and a stable 24x7 running app able to make it suitable for both, enterprise and residential use cases.

Sippo Web Collaborator is a multi-device application, so you can obtain mobility without renounce to the comfort of a laptop web client.

How it works



The Sippo Web Collaborator is a complete Angular HTML5 client that could be served for any web server. The collaboration features are implemented on the Sippo WAC or relayed to the WebRTC Gateway if a valid SDK is available. For audio and video calls, the signaling and media are sent directed to a generic gateway.

So the call is orchestrated by the WAC but the streams go browser to browser or browser to VoIP networks indistinctly.

This architecture provides a complete integration with existing systems, since the fingerprint of the client itself is minimal, so a highly customizable experience is guaranteed.

Key features:

- Works with Sippo WebRTC Application Controller (WAC).
- Supports unified communications features like audio and video calls, advance instant messaging, contact list.
- New features on v2.4: Whiteboard, Google contacts, feedback forms, complete full angular and HTML5 web interface.
- Implements all the necessary abstractions in order to transparently accommodate changes or specifics at the underlying WebRTC W3C API level (browser implementations , current discussions regarding WebRTC 2.0 API, ad-hoc implementations by browser vendors).
- Also available as application for smartphones, including iOS and Android and push notifications.

How it works:

- Supports browser to browser Communications and interconnection to legacy VoIP phones.
- Provides easy integration with existing Web service infrastructure.
- Remove VPN / network problems for remote users.
- WebRTC gateway abstraction.
- Integration with PBX, ACDs and VoIP infrastructure.

Main features

 **Audio & video**

 **Live chat**

 **File transfer**

 **Screen sharing**

 **Co-browsing**

 **User management by administrator**

 **Contact list**

 **Meetings**

 **Browser detection**

Audio and/or video calls

High voice quality. Supports G.711 and variable-rate OPUS codec. Supports VP8 and H.264 codec. Full screen mode available.

Contact list and agenda

Provision of default or per-user contacts. Customize personal data and integrate with AD and Google Contacts.

IM and group chat

Rich chat features like presence, typing detection, file sharing and file transfer. Chat rooms to host groups. MCU integration to support room groups.

Co-browsing experience

Possibility to share your screen, share an image and draw over it or to fill a form simultaneously with your remote party.

Call recording

Supports call recording of audio and video with legal compliance. Legal interception is possible with the use of third-party WebRTC to SIP media gateways.

Call detail records

Remote storage of CDRs with full call and contextual information. The WAC CDRs are exposed by a service API for BSS integration.

Statistics

Notifies activity for remote storage of platform usage statistics (active calls, users online, KPIs, etc).

Presence

Active real-time presence update on your contact list.

File transfer

Rich data files transfer experience thanks to the out-of-band data access, the integration with IM tool and the progress bar indicator.

Authentication

Multiple authentication backends including OAuth and OpenID, delegation on IMS/NGN, or digital certificate methods. Support to auto-login.

Provisioning

User provisioning including gateway authentication hide and integration with OSS systems. Multiple feature user roles and WebRTC gateway token based authentication.

Branding

Full capacity to define ad-hoc colors, logos, user interfaces and user privileges.

Meetings

Invite external guest users, integrate with iCal and contact them on the same app. SMS notification.

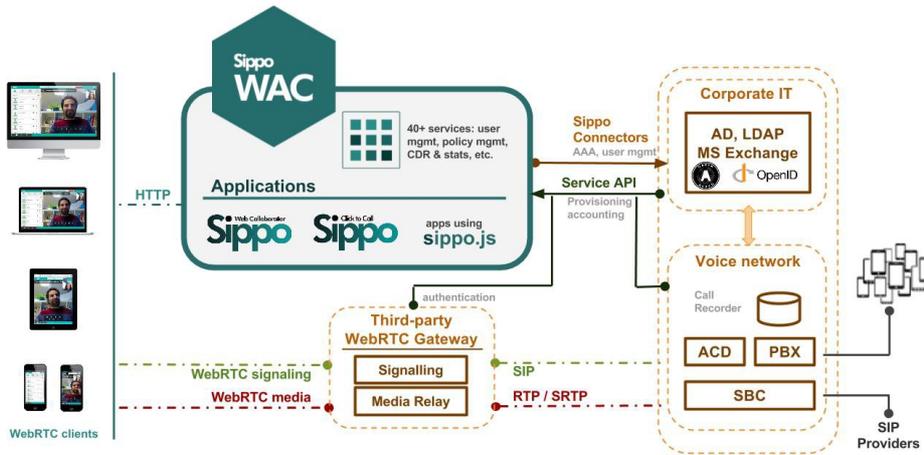
Contextual indormation

Remote party information: browser, geolocation, language, caller URL, etc. And third party tokens for integration with external apps.

Main features:

- Rich communications suite that support different unified communication methods with no need to install or update anything in the user devices.
- Provides abstraction forms to fill data with customers.
- Features to manage seamless integration with existing NGN, IMS or VoIP networks of services providers or enterprises.
- APIs to manage user provisioning, OSS and BSS.
- Support of different media gateways manufacturers to make Sippo interop with legacy networks.
- Carrier-class high availability solution with fully standards-based features.
- Shared forms to fill data with customers on real-time.
- Agent feedback forms to be launched automatically when call ends and include data into campaign statistics.

Architecture



The Sippo Web Collaborator works orchestrated by the Sippo WAC. The WebRTC communication is vendor-independent (15+ vendors supported), as all the status, data and advanced features are managed by the WAC. The WAC could be fully integrated with the enterprise network, providing a rich and advanced communication experience.

Architecture :

- 3GPP standard architecture. Where Sippo WAC plays the roles of WWSF and WAF.
- Designed for enterprise or telco networks.
- Highly scalable and HA design.
- Fully interconnected with existing assets for user management and service enablement.

DISCOVER other Sippo family applications

Sippo WAC

Sippo WebRTC Application Controller, WAC, is a solution that allows enterprises and service providers to deploy WebRTC applications fully-interconnected with their existing services (AD, OSS, BSS, etc.) and legacy VoIP or UC systems.

Click to Call Sippo

Sippo Click to Call is a light WebRTC endpoint designed for sporadic users who visit a website. It includes features of Sippo but adapted to multiple case of use.



Pol. A Granxa. P260
36400 O Porriño (Spain)

T: +34 986 911 644
W: www.quobis.com
M: info@quobis.com

Quobis Networks SLU. 2016. All rights reserved. Quobis, Sippo, IdentityCall and related marks are trademarks of Quobis. All other brand names are registered trademarks of their respective companies.

The content of this document is for informational purposes and could be changed by Quobis without notice. Quobis assures no liability resulting from technical or editorial errors or omissions, or for any damages resulting from the use of this information. Quobis has no obligation to develop or deliver any future release or upgrade or any feature.