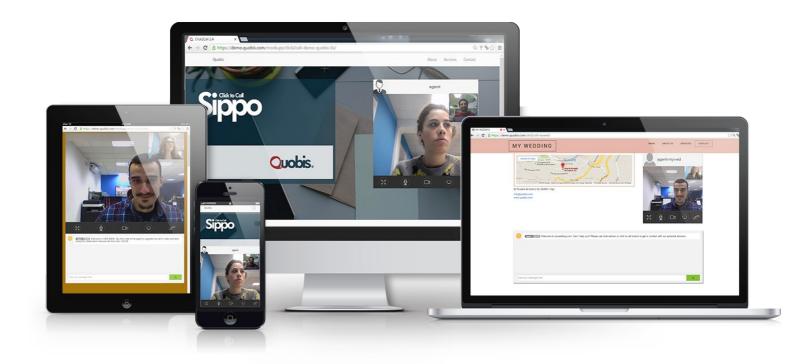


Sippo Click to Call







Overview

Sippo Click to Call is a fully-featured WebRTC-enabled click-to-call button with advanced capabilities that works with Sippo WebRTC Application Controller (WAC). Built for the web with our telco background knowledge, Sippo Click to Call allows IT managers of enterprises and service providers to deploy a lightweight click-to-call button.

It provides a easy integration for video integration for customer contact solutions, from full development to simple plug-n-play jQuery plugin. Sippo Click to Call provides audio, video, chat, file transfer featurers, whiteboard, ... but in addition to other solutions, it can be integrated with pre-existing networks, be scheduled for contact time-tables, regresion to legacy Call me Click To Call solutions, campaigns and agent assignation, form feedback for customer and agents, ... A complete solution for real scenarios.

How it works





Call is transferred to contact center and an agent is assigned (by WAC or pre-existing ACD)



Agents can use their legacy solution or move to web add video, chat, etc.



Sippo Click to Call is a jQuery plugin dowloaded from the Sippo WAC. This concept allow scheduled options or different deploys for each page or scope.

Once the customer clicks on the Click To Call button, the WAC will resolve the destination if ACD features are enabled (campaigns or hunt group). The call flows from the Click To Call WebRTC client to the WebRTC client of the agent.

On scenarios with VoIP integration (voice and video forking) the call goes to the WebRTC gateway, from it to an optional ACD that will distribute the call among the available

Key benefits:

- WebRTC based click-to-call button.
- Works with Sippo WebRTC Application Controller (WAC)
- Supports audio and video calls and/or advance instant messaging. Allow audio-video forking.
- Available in English, Spanish, Portuguese, Japanese and French.
- Customizable. Branding using easy CSS styles. Custom layout.
- Tested with industry leading SBCs and WebRTC-gateways.
- Per active endpoint and per concurrent session licensing. Also available as OEM.

How it works:

- Customizable per-customer scenarios.
- Rules (schedules, based on browser, geolocation...).
- Dynamic destination assign: Campaign and agent groups.
- Easy video integration on your existing audio network using the audio/video forking feature.





Main features



Audio & video



Live chat



File transfer



Screen sharing



Co-browsing

Audio or video calls

Optional video or audio only calls. High quality media supporting G.711 OPUS VP8/VP9 and H.264. Media selector on customer side

IM and chats

Chat-only option available. Chat features like presence, detection of typing, web links detection and chat history associated to each user. Proactive chats (without customer call needed).

Desktop sharing

Possibility to share your screen (with a small PIP of your video). Best option for technical call centers and training.

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

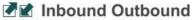
Remote storage of CDRs with full call and contextual information. The WAC CDRs are exposed by a service API for BSS integration.

Statistics

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

Context information

Remote party information: browser, geolocation, language, caller URL,





Contextual info



Analytics & stats



Call routing

S Browser detection

etc. Third party tokens for integration with external apps, and exposed Service API for CRMs.

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: Anonymous (C2C non privileged users), OAuth, OpenID (federation of identities with Google), LDAP integration, delegation on IMS/NGN, or digital certificate. Captcha support.

Provisioning

Database with user credentials for ad-hoc, automatic or federated with OSS provisioning. User policies.

Agent allocation

Call routing and ACD based on WAC information: presence, contextual data, campaign, hunt groups, etc. Dialpad with DTMFs.

Branding

Full capacity to define ad-hoc colors, logos, user interfaces and user privileges. WUI to manage branding. Possibility to access to OEM licenses.

Key specs:

- Rich click-to-call solution based on webRTC to provide a complete communication experience to visitors of a corporate site.
- Features to manage seamless integration with existing VoIP networks or legacy PBXs of contact centers and service providers.
- APIs to manage contextual information, OSS and BSS.
- Support of different media gateways manufacturers to make Sippo Click to Call interop with legacy networks.
- Carrier-class high availability solution with fully standards-based features.
- 3rd party contact center applications integration. Including call traceability and full user interface customization: distribute and modify the elements layout or use you custom css.
- Whiteboard: share an image and draw over it. Interact with your customers and help to understand shared documents.
- Customer / agent feedback forms Feedback information stored into the WAC and exposed by a Service API for Service assurance systems.
- Highly customizable solution, from user interface (CSS, HTML) to the user and agent media capabilities (from chat only to the full WebRTC communication experience.

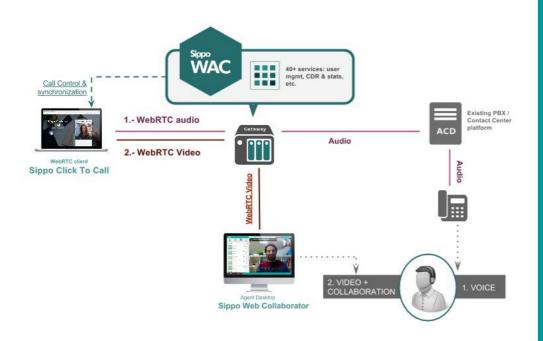






Voice and video forking

NEW!



Optional functionality that allows to make a call from the Click To Call button, one for audio via PSTN to the ACD of the call center. Optionally, this call could be upgraded to another browser to browser where could be selected video or any collaboration tool, like chat, screensharing, shared forms, whiteboard, file sharing, etc keeping always the audio into the VoIP phone extension.

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.



Sippo Web Collaborator is a multi platform WebRTC endpoint designed for enterprise or residential users that enables the access to multiple UC services.

- The best integration way for collaboration tools into your existing VoIP platform.
- Not all call centers are ready for video agents, but a full set of collaboration tools, from chat to screensharing is available.
- Keep traceability of your calls integrating a 3rd party token into the context information.



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