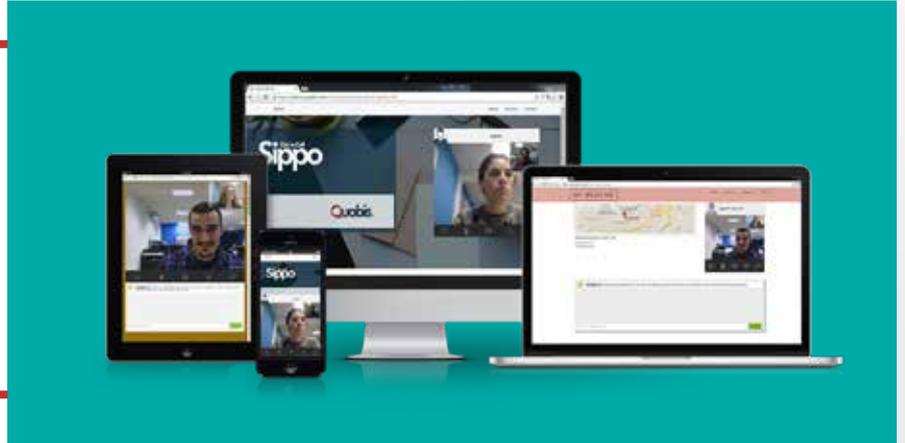


# Sippo .hub

Enterprise-grade server that enables companies with **customer-care applications** based on multimedia real time communications to defeat the challenges of digital transformation while keeping the **existing call center infrastructure**.



## // Add video and collaboration to call centers

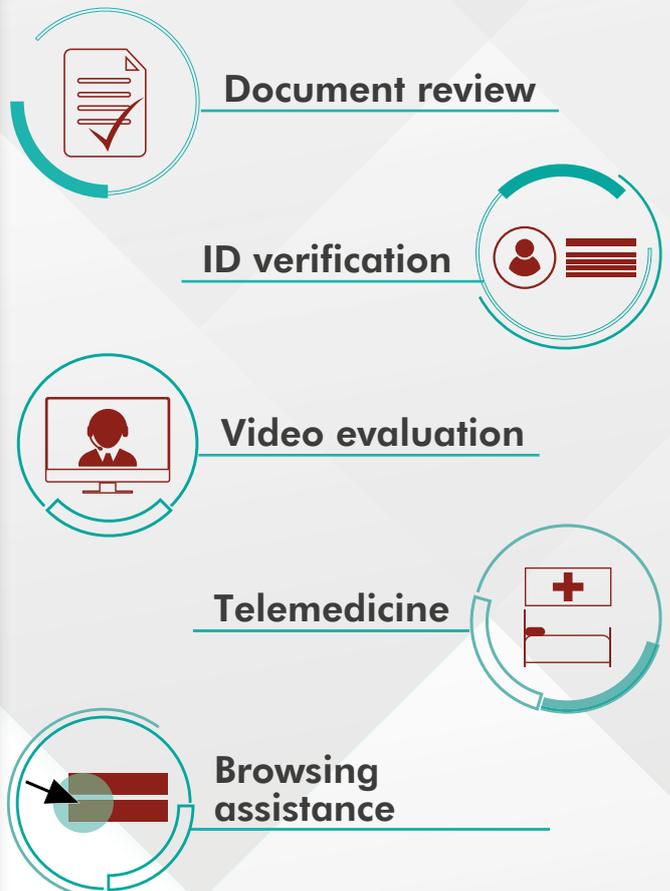


Sippo calls are adapted to SIP platforms and routed by the ACD. The agents can answer the voice call using the existing platform.



While keeping the voice call via the VoIP platform, video and collaboration (screen sharing, etc) will be placed browser to browser, adding multimedia capabilities to agents really fast.

**Sippo hub** provides modules for prebuild use cases and an open API/SDK to support and create new ones.





# // General features

## File transfer

- ➔ Send files between peers.
- ➔ Accept / reject file transference.
- ➔ Store the files transferred on the server side.
- ➔ Cancel on-going transferences.

FEATURE

## Presence

- ➔ Automatically / manually set user specific presence status.
- ➔ Retrieve user specific presence status

FEATURE

## Adaptive architectures

- ➔ On premises deploy over hardware host, VMware, docker, Azure, AWS.
- ➔ Adaptive architecture based on services and backends.
- ➔ Multiple feature implementations (backends) for the same service (feature).
- ➔ Configurable access to network interfaces.
- ➔ Integrated HTTP static server to serve the Sippo Apps integrated within the WAC or 3rd party apps.
- ➔ Support HA architecture based on cluster deployment.
- ➔ Multi Tenant support based on domains.

ARCHITECTURE

## Chat service

- ➔ Chat services for 1-to-1 or n-to-n based on WAPI messages.
- ➔ Chat service for 1-to-1 based on SIP instant messaging.

FEATURE

## Datchannels & datapipe

- ➔ Use datachannels.
- ➔ Use datapipe to share custom information between apps.

FEATURE

## Call routing services

- ➔ Notify a single user or group about an incoming call.
- ➔ Notify an external service about an incoming call.
- ➔ Check external REST API services to assign a call to an specific user.
- ➔ Route a call based on the specific WAC username.
- ➔ Route a call based on the user friendly name (local agenda).
- ➔ Route a call based on the IMS credentials username (SIP address)

FEATURE

## Recording capabilities

- ➔ Record 1-to-1 and group calls.
- ➔ Export recording files to an external storage server.
- ➔ Store partial recording of a call.
- ➔ Store events and timestamp marks on the recording.

FEATURE

## Push and notification service

- ➔ Wakeup a background app to receive an incoming call.
- ➔ Notify an app about other communication events (chat, file transfer, etc).
- ➔ Noify external users about a meeting using SMS or email.

FEATURE

## Scalability

- ➔ Handle up to 70k concurrent sessions and media calls (registered users).

ARCHITECTURE

MANAGEMENT

## Authentication & login

- ➔ Validate requests from anonymous users, user-pass pair, custom token and others.
- ➔ Create sessions based on user-pass schema, custom token, OAuth2for MS Active Directory, Google, Facebook, etc.
- ➔ Create sessions based on digital certificates pre-installed into the browser's device.
- ➔ Grant login, secure credentials and avatars to system users.
- ➔ Grant secure credentials for system users.
- ➔ Generic key-value pairs for a system user

MANAGEMENT

## User management & policies

- ➔ Assign capabilities to specific users, domains, applications, etc.
- ➔ Handle specific user/domain contacts.
- ➔ User's NAB (Network address book).
- ➔ Retrieve contacts from device.
- ➔ Handle contacts from a MS active Directory server.

MANAGEMENT

## Management services

- ➔ Admin CLI for authentication management.
- ➔ Admin CLI for user profile and capabilities management.

## Data exposition (SAPI & WAPI)

- ➔ Secure interface (WSS) of communication between SippoSDK apps and the Sippo server.
- ➔ Per feature endpoint permission customization.
- ➔ Secure interface Service REST API exposed from Sippo server available to be consumed by third party apps (SAPI).

## Activity logs and monitoring

- ➔ Session logging: user registration timestamp and presence status.
- ➔ Call logging: callerID, calleeID, timestamps, end reason, Qos (packet loss, jitter, codecs used), etc
- ➔ Log stack messaging (WebRTC signaling).
- ➔ Log WAPI messages (app signaling) and SAPI messages (management commands: CLI & 3rd party).
- ➔ QoS alarms: notifications per packet loss treshold.
- ➔ SNMP system monitoring support.



## // Integration with enterprise assets



Prebuilt applications for users (click to calls,...) and agent tools.



Ad-hoc development based on SDKs

# Sippo .hub



### CONTACT CENTER PLATFORM

Interconnection with ACD of the contact center, so customer can use the routing engine for allocate the available agentes and monitor service and KPIs.



### BUSINESS TOOLS

Service API helps to manage users, policies, services and retrieve information of Sippo from business tools (CRMs,...).

## // Sippo products



Fully-featured WebRTC-enabled unified communication suite with advanced capabilities.



WebRTC orchestrator that hides all the complexity in real field deployments of the technology.



Telco services in a mobile-focused easy-to-use application for residential and enterprises.



Quobis is a leading European company in the delivery of carrier-class unified communication solutions for telcos and enterprises.

Quobis is well-known as one of the leaders in the deployment of WebRTC technology after being involved in the release of the industry-first application server, called Sippo WebRTC Application Controller.

Today, this element is part of the core network several telcos worldwide. In addition Sippo is helping enterprises to defeat the challenges behind digital transformation processes using real-time communications

The company is headquartered in Vigo, Spain with partners throughout the world.