JOIN

A complete OTT client framework for desktop and mobile devices
Join is a complete VoIP client framework solution enabling service providers to offer next generation OTT services like Facetime, Whatsup, Viber and others.

The pre-integrated communication client, based on standard protocols, allows customers to focus on the user experience, GUI and add-ons, without having to know the complexities of VoIP ecosystems.

We provide our expertise and years of experience in VoIP software development to help you build a great customized VoIP app tailored to your client base.

Target Markets

SERVICE PROVIDERS – OTT RETAIL OFFERING

SERVICE PROVIDERS FOR ENTERPRISE HOSTED UNIFIED COMMUNICATION/PBX SERVICES

CORPORATIONS LOOKING FOR AN INTERNAL, SECURE COMMUNICATION SOLUTION

PBX MANUFACTURERS

WHY YOU SHOULD ROLL OUT IN-HOUSE OTT SOLUTIONS AND JOIN THIS MARKET?

$ 53.7 B

OTT communications MARKET VALUE

UK-based research house mobilesquared forecasts that the Over The Top (OTT) communications market is set to be worth $53.7 billion and have 2.1 billion smartphone users communicating by OTT services by 2017.

2.1B

smartphone users communicating by OTT SERVICES in 2017

For more information please contact: sales@voipswitch.com

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Why base your custom VoIP application on Join?

The market is fragmented and broad.

Every few months new versions of operating systems are released, and with each version there are changes in the device’s API, access to audio drivers and other pitfalls.

Join is always kept up to date and tested against developer beta releases.

Android is fragmented; there are devices from different vendors with their own modifications added on top of the official android releases. They may differ in many aspects, causing quality issues with audio and video across devices. Therefore a developer needs to test on particular devices without relying on compatibility with the system only.

Join is always tested on a wide range of devices to which all the latest models are continuously being added.

Short time to market:
Complete client framework reduces development, integration and testing phase.

Multi Platform:
Mobile devices based on Android, iOS and Windows Phone. Desktop version for Windows PC and MAC.
Join - complete OTT client framework

**Why base your custom VoIP application on Join?**

**Unparalleled User Experience:**
Join utilizes an advanced mechanism for NAT and Firewall passing through. It supports ICE with STUN/TURN and TCP TURN. It just works from any network.

**Highest media quality:**
Join always tries to establish a peer to peer media path while signaling is sent through the SIP server. This approach is especially important in the OTT market with support for wideband audio codecs and high resolution video calling. Peer to peer greatly offloads the operator’s infrastructure and servers.

**Technical support and development assistance:**
2nd line of support, continuous updates of the app. Post deployment work on enriching the app by adding new, or modifying existing, features.

**Close work with our development teams:**
Our engineers will help you through the whole process, from the UE and GUI design to the most advanced integration with 3rd party platforms, until the app is uploaded to your appstore accounts.

**Hardware integration:**
Join is optimized for various chipset and audio drivers of different manufacturers. It also integrates seamlessly with other peripherals such as cameras, Bluetooth etc.

**Interoperability:**
Join is standard based and has been tested with all popular softswitches and IMSes in the market – Huawei, ZTE IMS, Broadsoft, Nokia Siemens Networks, PortaOne and many others. In addition, we work closely with PBX manufacturers like LG Ericsson, delivering them custom FMC VoIP clients to work with their equipment.

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Video

Starting off as a novelty, video calling has quickly been adopted by users and is now an expected part of any OTT client.

This transition was possible thanks to the growing internet bandwidth, new wireless technologies (3G, 4G, LTE) and significant improvements in video encoders.

Supported video codecs:

- **VP8** – the codec used by Skype, Google Hangout, WebRTC

- **H264** – codecs backed by GSMA RCS specification

- **H263** + and **H263 older codecs** still used in some specific deployments

Supported video resolution

- **QVGA, CIF, VGA** and **HD** (980x720, dependent on a device’s specs).

Up to 30 frames per second.

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Multiparty Video conference

Mixing on the device’s side. This feature is currently supported only in desktop versions. Allows up to 5 video participants in one conference.

Audio HD codecs

Join supports a whole range of audio codecs, from narrow band designed to deliver smooth voice even in harsh network conditions to the latest High Definition wideband achievements in the industry.

Select codecs: G729, G722, G711, GSM, Speex, SILK, OPUS, AMR WB and NB

OPUS codec – unparalleled voice quality, choice of google hangout and WebRTC project. Required for interworking with WebRTC browser based clients.

Voice processing components: Echo cancellation, Noise Reduction, Packet Loss Concealment.

Quality Indicator.

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**Why peer to peer?**

The bitrate needed for decent quality video conversation is still many times greater than for audio calls, which imposes a big burden on the operator’s network. Peer to peer communication also helps quality when the users are in the same region and closer (better connected) to each other than when routed through a media server. The peer to peer path thus eliminates latency problems without deploying relay servers in multiple locations.

The mechanism used for establishing a peer to peer connection is ICE - a framework using TURN (and its extension STUN).

ICE is used also in WebRTC and services like Facebook.

In the event of a failure in establishing p2p ICE automatically falls back to TURN – connectivity through a relay server.

**Nat or firewall? ICE yet again.**

Traversal Using Relays around NAT (TURN) is the latest development. The best results are when used with ICE - Interactive Connectivity Establishment (ICE). If there is indeed one path for two clients to communicate, then ICE will find this path. And if there is more than one path, ICE will use the best/most efficient. Relay is only used as the last resort, thereby minimizing the bandwidth and latency issue of relaying.

TCP TURN support RFC 6062 for networks blocking UDP traffic.

Blocked areas – tunnel technology masking the traffic patterns from the filtering devices.

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**PUSH – always on**

When the application is closed (for example by user or by a system) all requests are routed through the PUSH mechanism.

Supported platforms: Apple iOS, Android, Windows Phone 8.

Supported requests: audio and video call, instant messages, voicemail notification, file sharing notification.

PUSH can also be used for sending bulk marketing or service messages.

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**Join - complete OTT client framework**

**PUSH NOTIFICATION**

- **user A**
- **INVITE**
- **user B**
- **INVITE with replaces header**
- **Home server**
- **EMC/API**
- **PUSH request**
- **PUSH notification**
- **Application closed**
- **user B**
- **User answers, the app wakes up**
- **PUSH messaging Cloud**

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Security

VoIP traffic can be intercepted on its way through the public internet, and calls can be eavesdropped and recorded. Confidentiality of information is essential for the enterprise market.

Security of communication is one of the main factors when choosing a Unified Communication platform.

Join addresses this concern by utilizing strong encryption technologies for texting and voice and video calls.

SRTP (RFC 3711) AND SRTP SDES (RFC 4568) FOR MEDIA ENCRYPTION.

ENCRYPTION FOR BOTH AUDIO AND VIDEO CALLS.

128 OR 256 - BIT AES ENCRYPTION.

OPTIONAL ZRTP SUPPORT.

TLS - SIGNALING ENCRYPTION, FOR ALL SIP COMMUNICATION INCLUDING INSTANT MESSAGING.

IMS/VoLTE, GSMA RCS

IMS/VoLTE and RCS standards support. Tested with various IMS equipment vendors.

Successfully deployed in IMS ecosystems based on Nokia Siemens Networks, ZTE, Huawei, Samsung.

KEY HIGHLIGHTS

- GSMA IR.92 & IR.94 specification support
- IMS registration, authentication and addressing
- VoLTE Voice and Video call establishment
- IMS Preconditions
- AMR WB & NB codecs
- H.264 video codecs
- Presence with Resource List (XDMS) support
- Instant messaging

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WebRTC interworking

Connect WebRTC based web softphone clients with your SIP infrastructure using the Join for WebRTC framework comprising JavaScript libraries and a web to SIP bridge.

Brilliant quality with support for ICE (peer to peer), vp8 for video and OPUS for audio.

Support for TURN as a last resort path for media (if peer to peer fails).

Full encryption for both audio and video.

VOIPSWITCH USER PORTAL

Voipswitch User Portal – fully WebRTC compliant html5 based portal.

Audio and Video calls from Portal to Join mobile and desktop clients.

Integrated Network Address Book, support for Presence RCS and Instant messaging (Chats).

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RCS services and beyond RCS

Support for Joyn GSMA Rich Communication Suite recommendations:

**Key features:** Auto provisioning over the air, Discovery via Options, File transfer, Geo location sharing, vCards, Enhanced Address Book, Conversational Instant Messaging (Chat), Social Presence Profile with portrait icon, Presence with support for RCS standard lists.

**Voipswitch RCS approach** - full suite of desktop and mobile clients together with the RCS Node platform which comprises all the elements required to run RCS services.

The platform integrates with 3rd party softswitches. Offered as standalone or in the cloud.

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Beyond RCS – Voipswitch extensions to RCS:

- Multidevice support.
- Find friends public directory.
- Multiple private IDs, e.g. phone number, nickname, email, Facebook or Google Plus ID.
- Voice and video encryption.
- High quality OPUS and VP8 codecs for onnet calls.
- Separate config for breakout (offnet) calls (e.g. different codecs, different sip credentials).
- Converged Address Book: network address book synchronizing contacts across different devices and platforms.
- DIDs (virtual numbers) support.
- Integration with self-care Portal with WebRTC audio and video calling features.
- Presence and Instant Messaging.
- Shared history of chats and calls/ voicemails across all devices.
- Visual voicemail with personalized greetings.

You can find more on RCS here:

http://www.voipswitch.com/products/sip-softphones/rcs/