VoipSwitch
Tomorrow's Communication, Today

JOIN for mobile - datasheet

Join for mobile

Cross platform client framework for building VoIP/RCS applications

Supported platforms: Android, iOS, Windows Phone 8, Blackberry X

Join for Mobiles is a set of a feature-rich VoIP clients developed by Voipswitch for the most popular mobile platforms. The clients are based on SIP and other open standard technologies supported by IMS and VoLTE. Its advanced underlying qualities complimented by the sleek design and the deep understanding of User Experience is what makes Join the market leader.

Join is a base for building white label fully customized communication application.

Join VoIP client is also a superior mobile dialer for enterprises as an extension to an IP PBX. An example of Fixed to Mobile Communication is the LG Ericsonn project where our client was bundled with PBX appliances.

Winner of the 2011 Product of the Year Award from Communications Solutions

JOIN GENERIC

Join generic version is also available as a demo on appstores; this version is intended to work with 3rd party PBXes and VoIP services as due to its universal purpose some features are not enabled (as they need certain server side integration). The available feature set remains extensive enough to feel the design approach and quality of the product.

Join generic supports multiple accounts; a user can be registered to multiple SIP servers at the same time.

Join for mobile

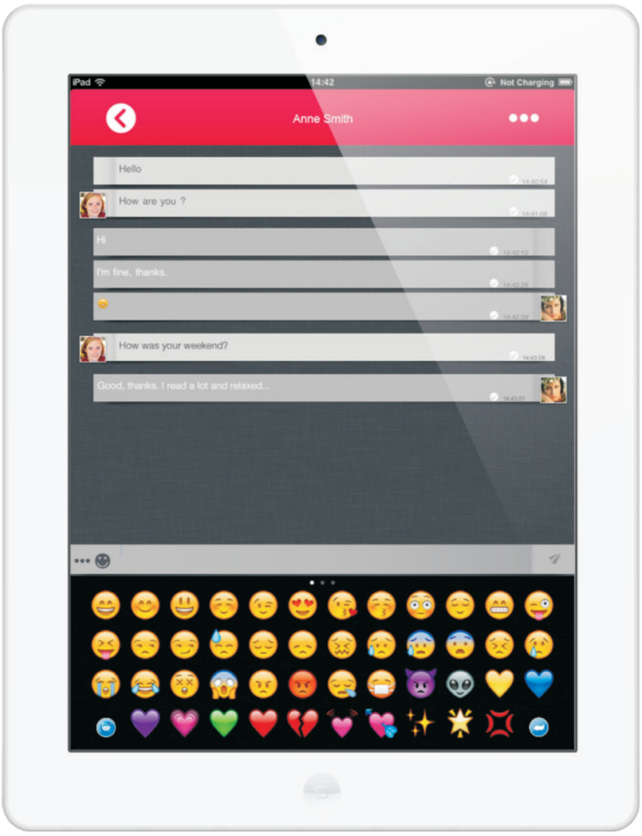
Key highlights

WebRTC compatible:
Call from UC web based solution directly to native iOS or Android SIP client.

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

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

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.



Enriched communication

Join allows high quality audio and video calls. Users can send instant chat messages and share multimedia and geolocation during a call.

In addition you can see the status of your buddies using various implementations of the SIP PRESENCE mechanism.

Excellent quality with VP8 video codec or h264. Audio codecs include the latest achievements such as OPUS with all preprocessing ensuring unrivaled user experience.

Join for mobile

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

Security

Necessary for business VoIP solutions. Support for the strongest encryption standards used in real time communication.

Secure SIP signaling, Presence, Instant messaging and media including audio and video calls.

Branding

The branding in its basic form always includes changing the name of the application, preparing binaries to be uploaded to the customer's account on Apple and Google Play stores, adding the logo and splash screen

Further phases of branding usually encompass changing the color schemes and look of particular screens. If you want a completely unique look and feel we can fully customize

Our designers are happy to work with our customers on beautiful and innovative graphical interfaces combining a great user experience with an esthetic and modern look.

For more information please contact: sales@voipswitch.com

www.voipswitch.com

Join for mobile

Technical specifications

General features

Multiple account support
Active registrations to different server.

Multiple calls facility
Swipe to switch to another call.

Background operations support

PUSH notifications
On incoming calls and messages.

Multiple languages

Protocols

- Full SIP compliant
- IMS/VoLTE extensions support
- Presence with Resource List, Watcher Info XCAP for XDM
- STUN/TURN and ICE framework (peer to

Security

- SRTP encryption for audio and video
- AES with 128 or 256 bit key
- TLS encryption for signaling

Video

- WebRTC engine (Android)
- Codecs: VP8, h264, h263
- Resolutions from CIF to HD (subject to device's performance),Frame rate up to 30 fps
- Congestion control

Messaging

- Instant Messaging SIP SIMPLE (OMA), CPM, CPIM
- IM large messages (MSRP)
- Deferred messages
- SMS over IP
- Converged inbox

Audio

- Preprocessing algorithms including Echo Cancellation, Noise suppression.
- WebRTC media engine (Android).
- Wide range of codecs: OPUS, SILK, AMR WB and NB, g722, Speex, iLBC, g729, GSM, g711

VoLTE/IMS

- GSMA IR.92 & IR.94 specification support
- IMS Registration, Authentication and addressing
- VoLTE Voice and Video call establishment
- IMS Preconditions

RCS

- **Voipswitch RCS extensions support**
- **GSMA RCS standard compliant**
- **User Discovery using Options, File transfer**
- **Provisioning**