WebRTC stands for Real-Time Communication and is a new technology being drafted by the World Wide Web Consortium (W3C) and IETF groups. This technology has the ambition to bring native real-time features (audio, video and arbitrary data) to the web browsers without requiring additional plugins.

SIP stands for Session Initiation Protocol and is a signaling protocol defined by the IETF in RFC 3261. SIP is widely used today to manage VoIP (Voice over IP) communication sessions and has been chosen as signaling protocol for Next Generation Networks such as IMS (IP Multimedia Subsystem) or LTE (Long Term Evolution). The protocol has quickly become the de facto standard used to interconnect the IP world (Internet) with the PSTN (circuit-switched telephone networks).

webrtc2sip is a smart and powerful gateway using RTCWeb and SIP to turn your browser into a phone with audio, video and SMS capabilities. The gateway allows your web browser to make and receive calls from/to any SIP-legacy network or PSTN. As an example, you will be able to make a call from your preferred web browser to a mobile or fixed phone.

**Scope**

This technical guide is a reference document explaining why you need webrtc2sip and how to leverage its power.

**Architecture**

The gateway contains four modules: SIP Proxy, RTCWeb Breaker, Media Coder and click-to-call service.

[Figure 1: Architecture]

The HTML SIP client is any endpoint implementing draft-ibc-sipcore-sip-websocket-06. We highly recommend using sipML5 which is known to work and provide good performances.

**3.1 SIP Proxy module**

[Figure 2: SIP Proxy architecture]

webrtc2sip – Smart SIP and Media Gateway for WebRTC endpoints

Source: http://webrtc2sip.org/technical-guide-1.0.pdf
Webrtc2sip Elements

Source: http://webrtc2sip.org/technical-guide-1.0.pdf
Layout of Today’s Demonstration

- **sipML5**
- **Doubango**
- **PSTN**
- **Desk Phone**
- **Mobile Phone**
- **SIP Proxy/Registrar**
- **webapp.js**
- **VP8**
- **opus**
- **webrtc2sip Gateway**
- **ICE**
- **DTLS/SRTP**
- **SRTCP-FB**
- **RTP**
- **RTCP**
- **G.711**
- **H.263**
- **VP8**
- **opus**
- **SIP Phone**
- **NICTA**
A Test Drive

sipML5 → Desk Phone
sipML5 → Mobile Phone
sipML5 → SIP echo test
sipML5 → X-Lite (SIP video call)

sipML5 → X-Lite (SIP video call)
sipML5 → Desk/Mobile Phone (callback service)