1. **WebRTC** (Web Real-Time Communications)

**Real-time communication for the web**

With WebRTC, you can add real-time communication capabilities to your application that works on top of an open standard. It supports video, voice, and generic data to be sent between peers, allowing developers to build powerful voice- and video-communication solutions. The technology is available on all modern browsers as well as on native clients for all major platforms. The technologies behind WebRTC are implemented as an open web standard and available as regular JavaScript APIs in all major browsers. For native clients, like Android and iOS applications, a library is available that provides the same functionality. The WebRTC project is [open-source](https://webrtc.googlesource.com/src/) and supported by Apple, Google, Microsoft and Mozilla, amongst others. This page is maintained by the Google WebRTC team.

1. In accordance with the WebRTC standard, Google Meet uses **VP8 and VP9 video codecs** for video stream compression and **Opus audio codec** for voice stream compression.
2. root@asterisk:~# apt-get install build-essential wget libssl-dev libncurses5-dev libnewt-dev libxml2-dev linux-headers-$(uname -r) libsqlite3-dev uuid-dev