Temasys

Embed Real-Time Communication

Any App I Any Device I Any Scale

Temasys

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WHAT IS WEBRTC

WebRTC

THE MARKET

2016
USD $10.71 bn

2025
Estimated
USD $81.52

Sources: Transparency Market Research, 2017

THE MARKET

6 Bn

No. of WebRTC enabled devices by 2019

2 Bn

Individuals using WebRTC by 2019

US$6B

TAM

Total Addressable Market

Sources: Disruptive Analysis, 2015 + Company Research

FACEBOOK

Sources: Facebook.com, Dave Marcus April 2017

GOOGLE

Google Allo

Google Duo

Hangouts

SNAPCHAT

CUSTOMER SPOTLIGHT

Webinar Jam

GENESISDIGITAL

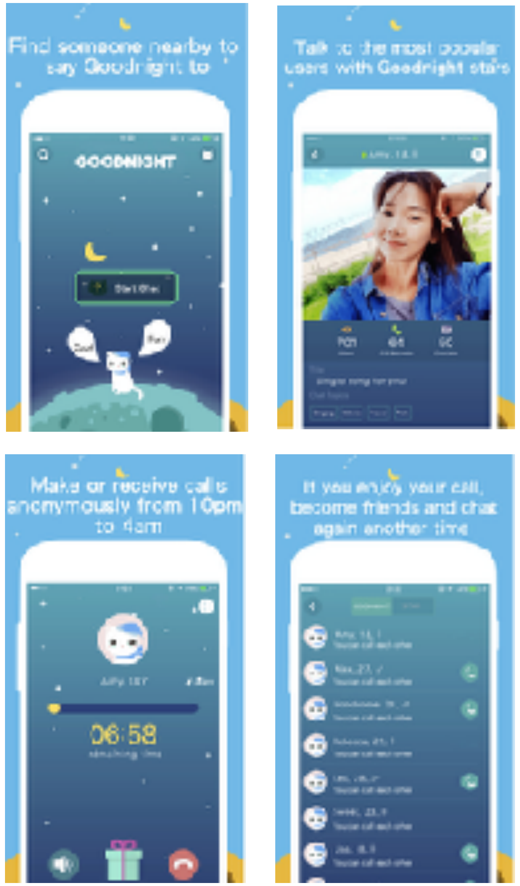
WEBINARjam

Genesis Digital, creators of the industry's premier live event platform, WebinarJam

The world's 2nd largest online webinar provider striving to overtake Citrix's #1 position – strong growth forecast.

CUSTOMER SPOTLIGHT

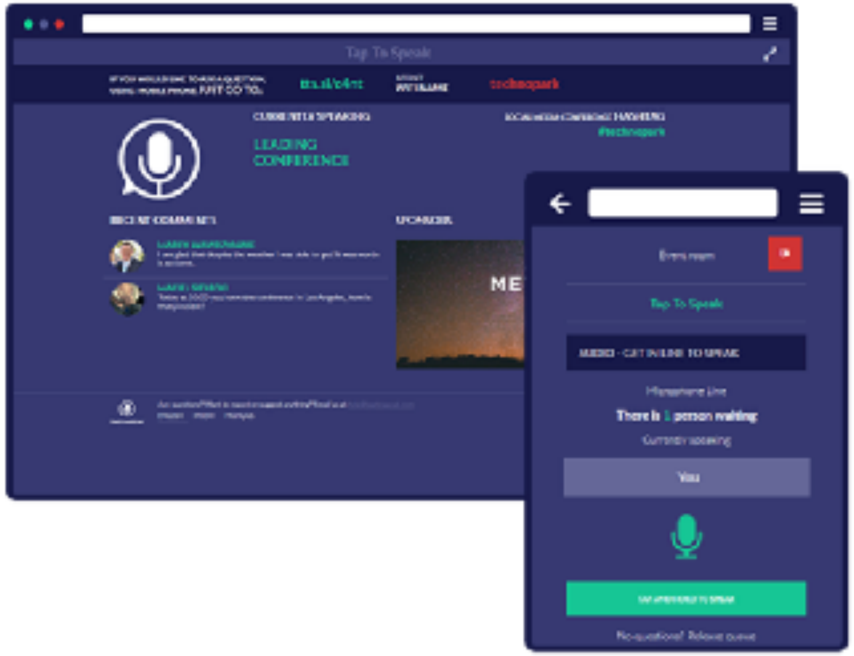
Paktor



Paktor's Goodnight social app delivers anonymous voice calling to iOS and Android devices.

- A 5-star rated social app available from Apple iTunes Store or Google Play.
- Strong growth in Taiwan, expanding into other Asian countries.

Tap to Speak



Tap To Speak is a Software as a Service (SaaS) company based in Phoenix Arizona and Lodz Poland, that provides a web-based tool through which live event audiences can communicate with event leaders and speakers in real time.

- The Tap To Speak app turns any smartphone into a microphone.

CUSTOMER SPOTLIGHT

Big White Wall

A safe online community of people who are anxious, down or not coping who support and help each other by sharing what's troubling them, guided by trained professionals.

- Available 24/7, Big White Wall is completely anonymous so you can express yourself freely and openly. Professionally trained Wall Guides ensure the safety and anonymity of all members.

WHAT IS WEBRTC

BORN FROM GOOGLE

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GLOBAL IP SOLUTIONS

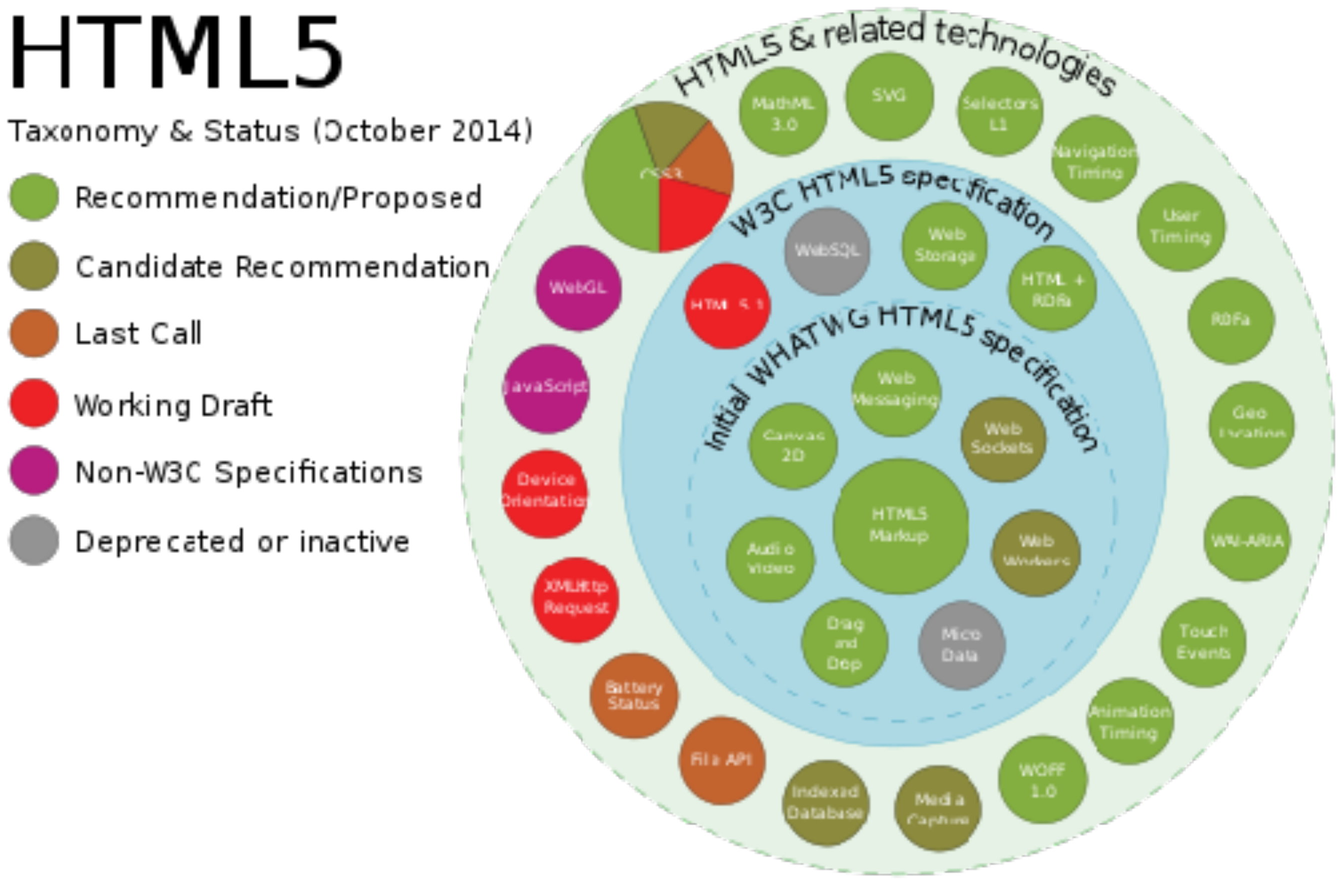
on2 technologies

Google

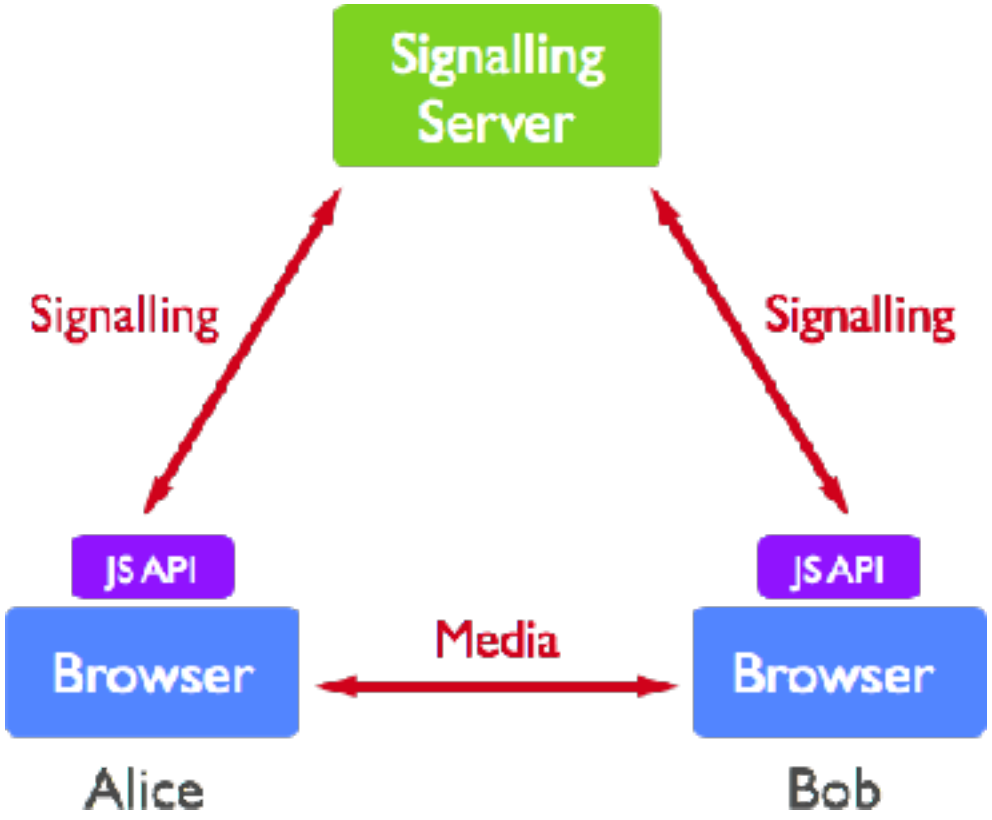
WebRTC

COMPLETELY STANDARDS BASED

BUILDING WEB TECH



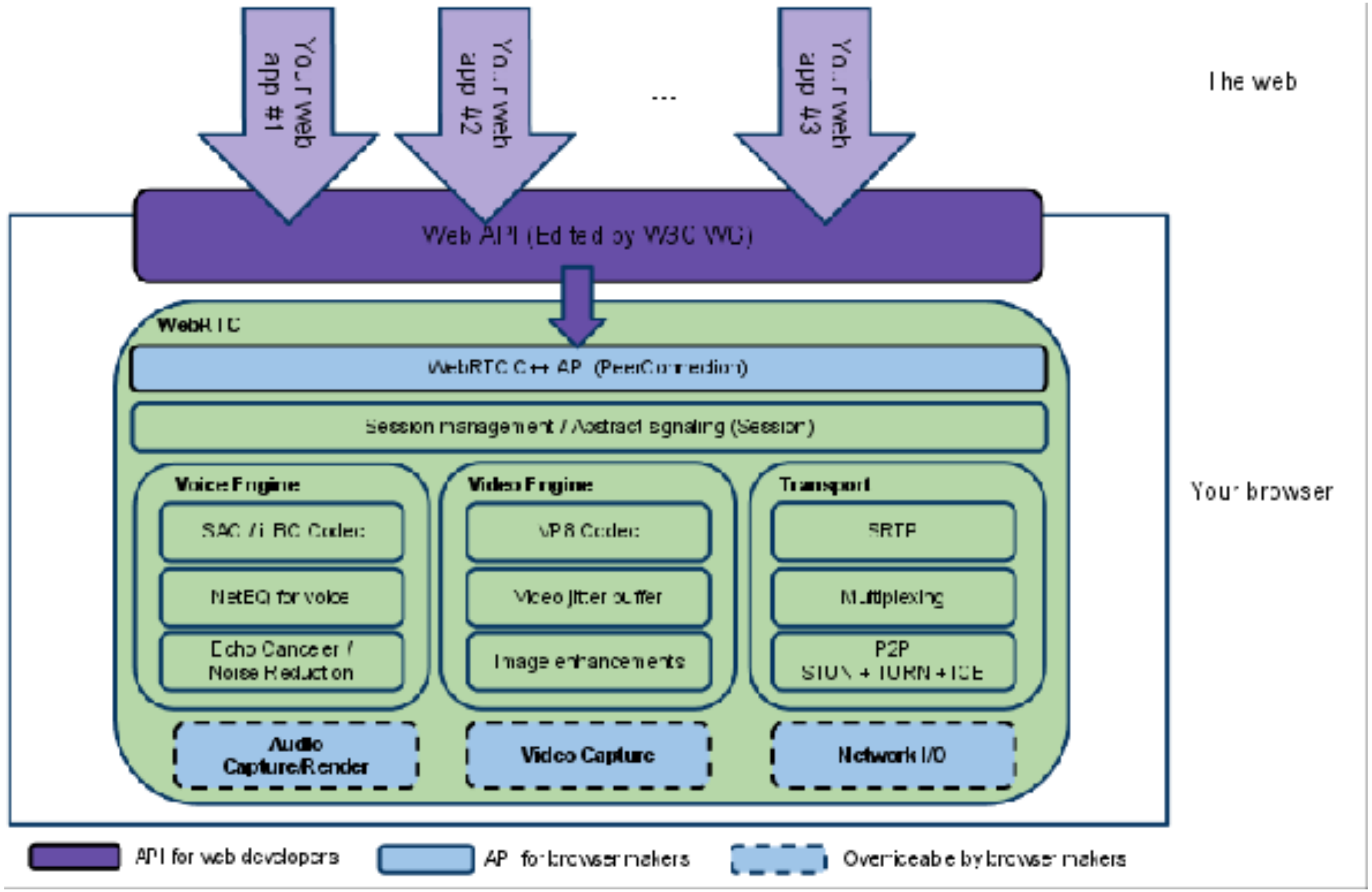
OK…SO WHAT IS IT



TECHNOLOGY RICH

- Web APIs
- Session Management with SDP
- AV Codecs
  - G7.11
  - Opus
  - VP8/VP9
  - H.264
- NAT Traversal Protocols
  - STUN
  - TURN
  - ICE
- SCTP for Data/File Transfers
- Transport Protocols
  - RTP/RTCP/RTCP-FB and RTCP Mux
  - Bundle
- Encryption and Security
  - DTLS-SRTP
- Audio/video Jitter Buffers
- Packet Loss Concealment
- Local Audio Processing
  - Audio Echo Canceller
  - Auto Gain Controls
  - Noise Reduction

WEBRTC CLIENT ARCHITECTURE

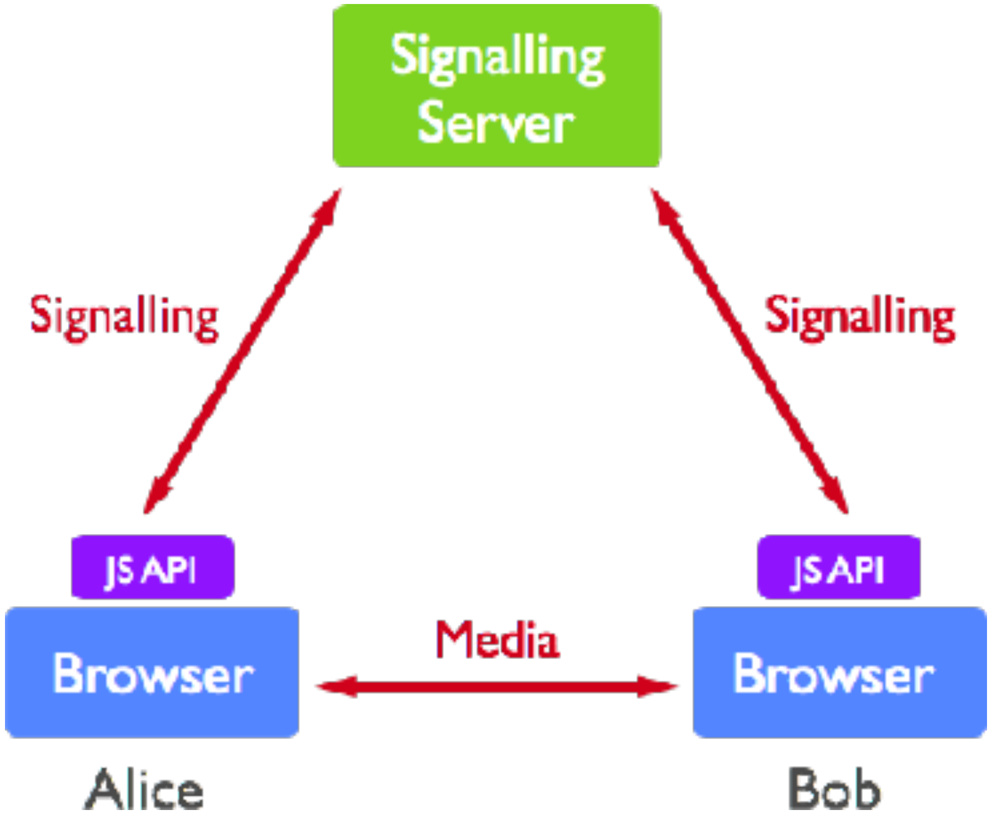


WEB API

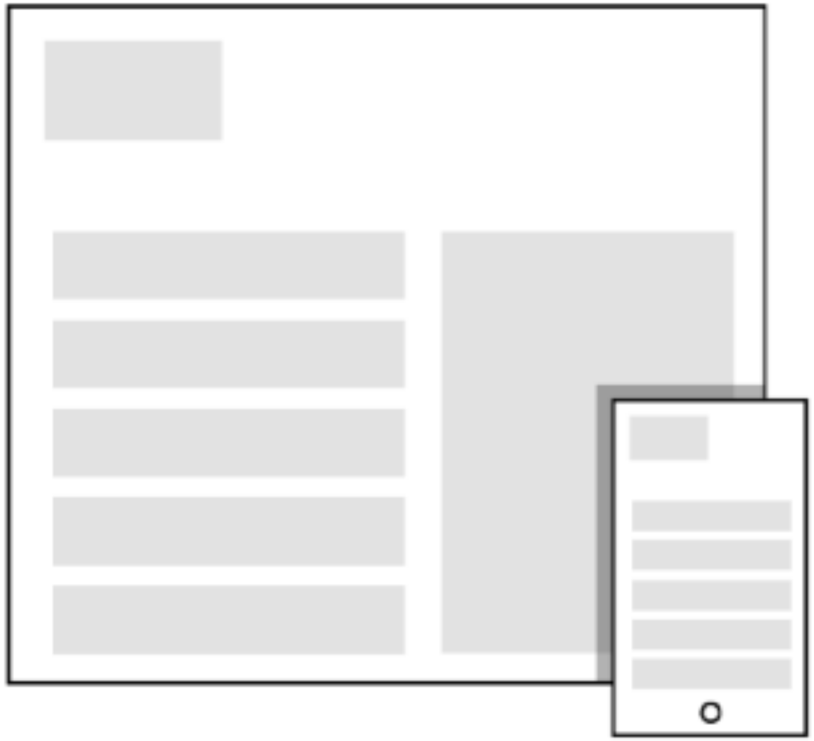
WebRTC provides the basics

- GetUserMedia
  - Access/Handling camera and microphone
- PeerConnection
  - Session/capabilities exchange and sending/receiving media
- DataChannels
  - Sending non-media between browsers

OK…SO WHAT IS IT



SO HOW DO I GET STARTED



STEPS

- 1. Get the Microphone and Camera – **GetUserMedia()**
- 2. Create a signaling mechanism for two browsers to connect (WebSocket and JSON?)
- 3. Create a PeerConnection (**RTCPeerConnection**)
- 4. Creating/Sending/Receiving Offers/Answers of Media
  - **RTCPeerConnection.createOffer()**
  - **RTCPeerConnection.createAnswer()**
  - **RTCPeerConnection.setLocalDescription()**
  - **RTCPeerConnection.setRemoteDescription()**
  - **RTCPeerConnection.onicecandidates()**

WEBRTC IS AN ENGINE

WEBRTC NEEDS INFRASTRCUTURE

BUILDING SUPPORTING INFRASTRUCTURE

- Define signaling protocol
- Create a signaling server
- Browser incompatibilities
- Mobile/Browser interoperability
- STUN/TURN server for fw/nat traversal
- Media server
  - Better bandwidth management
  - Media/Call manipulation
  - Recording

- Common API Layers
- Common Libraries
- Dev Ops
  - Redundancy
  - Scaling
  - Load Balancing
  - HA

APPROACH 1: BUILD FROM SCRATCH

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WebRTC About ▾ Developer ▾ Native Code ▾ Web APIs ▾

Home > Start

Getting Started

New to WebRTC?

Here are some suggestions to help you get started:

1. Get an overview of WebRTC: video, slides.
2. Find out more about WebRTC architecture and JavaScript APIs: Getting Started With WebRTC.
3. Try out our code samples and live demos.
4. Try our codelab.
5. Read through the code for the canonical video chat app appr.tc. The repo is at github.com/webrtc/apprtc.
6. For iOS, Android or the C++ WebRTC APIs, take a look at the Native APIs resources below.
7. Ask general questions on Stack Overflow, deeper technical WebRTC questions on discuss-webrtc.

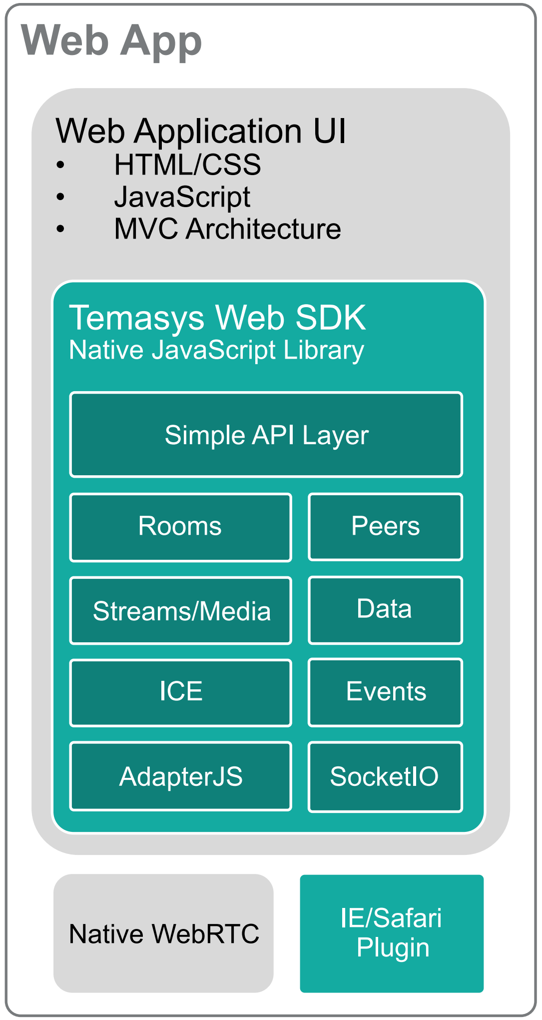
More resources below.

Contributions and updates welcome.

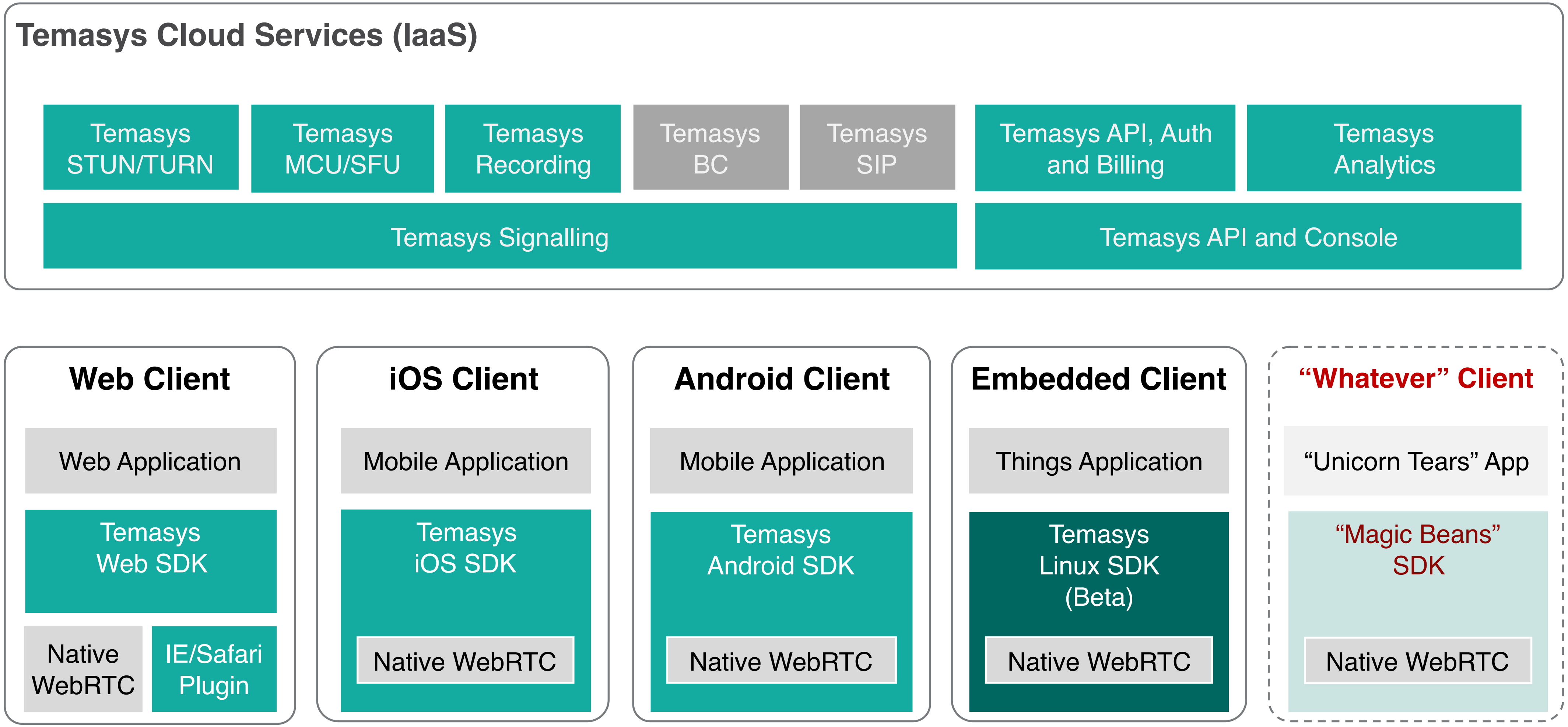
https://webrtc.org/start/

APPROACH 2: HYBRID WITH OPEN SOURCE

APPROACH 3: USE A PLATFORM



APPROACH 3: USE A PLATFORM



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Jump Start Your Prototype!

- Sign up for a Temasys account
- Set up your first app key
- Download our ready-to-use sample apps
- Sub your app keys into the sample app config file placeholders

https://github.com/Temasys

THANK YOU

*"The coming year promises to be the beginning of a sea change in communications, **the age of Communications Applications that have communications delivered directly in the application."***

"WebRTC is rapidly becoming the dominant VoIP (voice and video) protocol. Communications within applications is changing the landscape of both communications and applications."

Phil Edholm
Communications Industry Expert

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