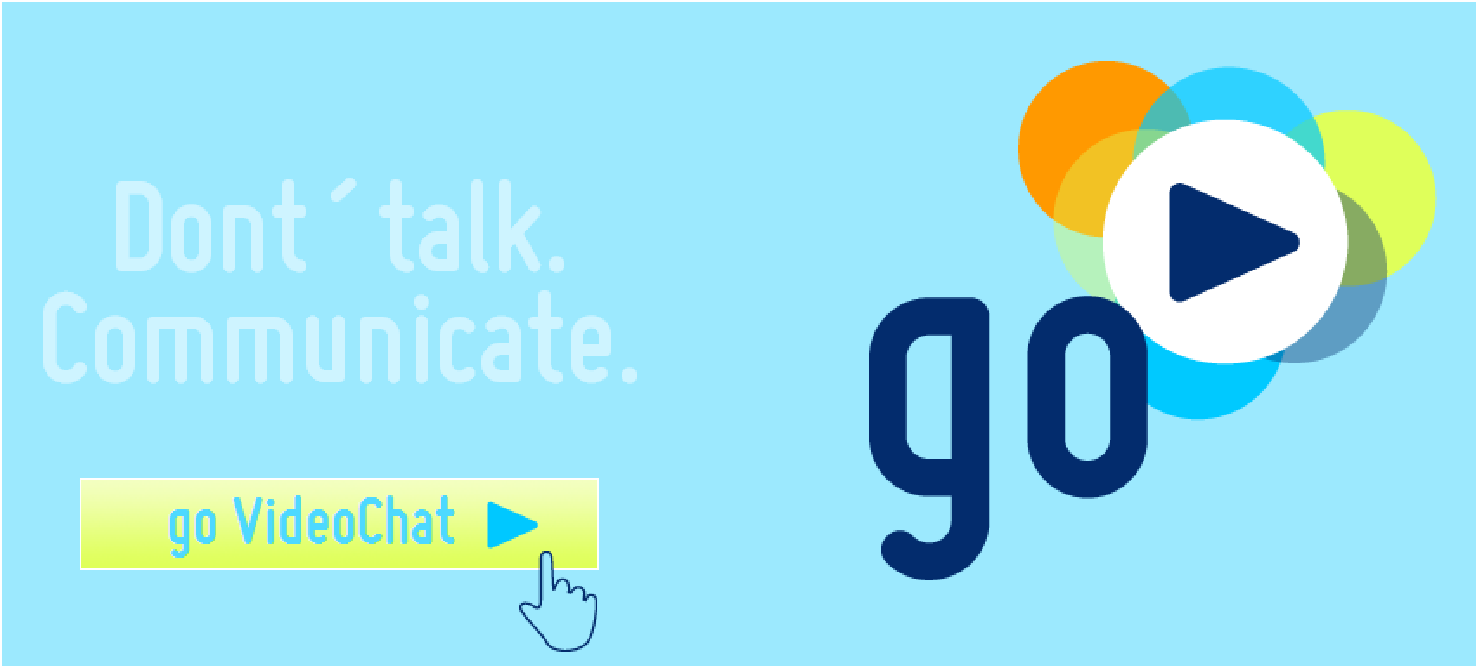
ESTOS

Communication Solutions



webRTC and XMPP

Philipp Hancke, XMPP Summit 2013

What is this webRTC thing …
…and why should XMPP developers care?

- I assume you know what XMPP is…
- … you might have heard of Jingle
  - the XMPP framework for establishing P2P sessions
  - used for VoIP, filesharing, …
- … you might have also heard about this webRTC thing
  - doing VoIP in the browser
  - without plugins
  - „no more flash“
- Do you want to know how it relates to XMPP ?

What is webRTC?

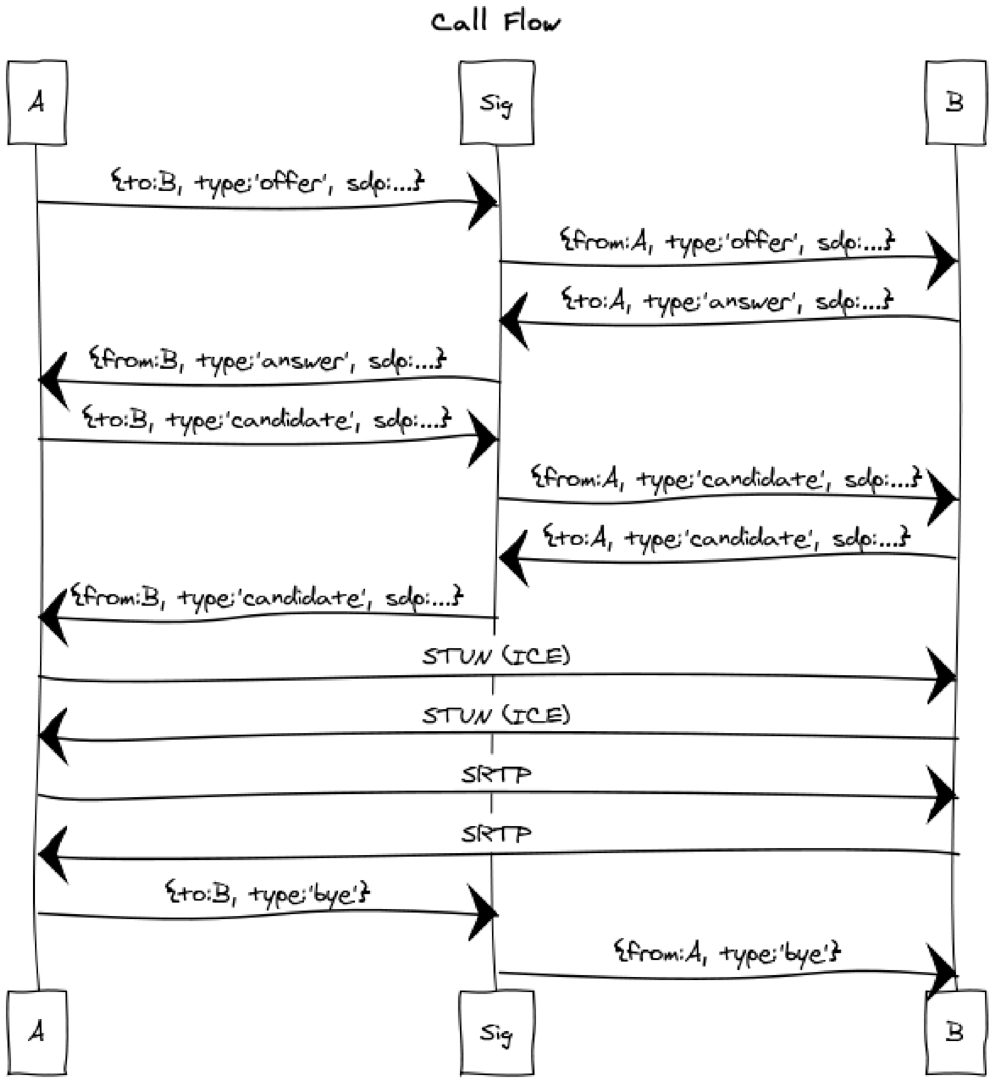
- P2P sessions between browsers
  - no servers involved in media transfer
  - using open standards
  - Javascript API in the browser
  - also an BSD-licensed C++ library from Google
- Want to know more?
  - Listen to the evangelists!
  - Justin Uberti http://www.youtube.com/watch?v=E8C8ouiXHHk
  - Jose de Castro http://vimeo.com/52510068
  - Cullen Jennings http://vimeo.com/cullenfluffyjennings/rtcwebexplained

Initiating P2P sessions

- initiate a P2P session between two browsers
  - negotiate media codecs, NAT traversal, etc
  - media is sent P2P

- you need a session initiation protocol
  - SIP?
  - JSEP?
  - H.323?
  - Jingle!
- webRTC does not mandate a signalling protocol
  - WG decision

Call Flow - JSEP

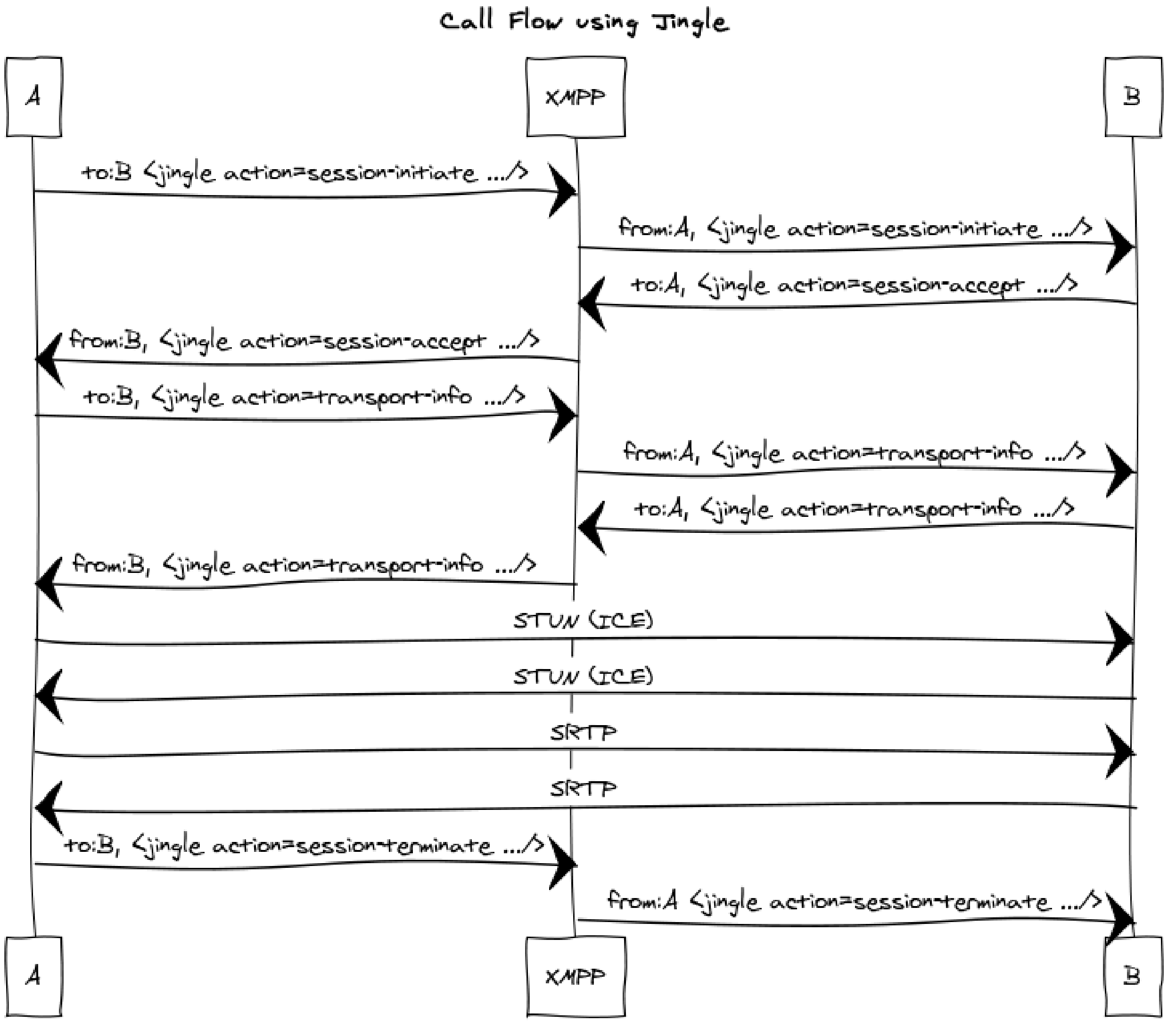


Jingle

- You can use Jingle as signalling protocol
- together with BOSH or XMPP over websockets in the browser
  - Demo later

- But…
  - webRTC uses the Session Description Protocol as an API
  - Jingle does not use SDP
  - You need a mapping SDP -> Jingle -> SDP
  - Complicated, but doable
  - Topic for breakout

Call Flow - Jingle



webRTC-Jingle usecases

- Browser -> BOSH -> XMPP Server -> BOSH -> Browser
- Browser -> BOSH -> XMPP Server -> BOSH -> native client
- Browser -> BOSH -> XMPP Server -> XMPP Federation -> …
- Browser -> BOSH -> XMPP Server -> SIP Gateway -> …

webRTC today

- Developing a standard takes time…
- webRTC was kicked off in October 2010
- But you can try webRTC today
- Google Chrome
  - Enabled by default since M23
  - Currently at M24 (opus codec)
  - Support for DTLS-SRTP in Canary (M26)
  - Uses webRTC library and libjingle
- Firefox
  - Nightly
  - Uses webRTC lib as media engine, does not use libjingle
- No interoperability between Chrome and Firefox
  - will change very, very soon
- No support in mobile browsers (yet, first signs of code last week)

webRTC today

- Get Chrome and test it at https://apprtc.appspot.com/
  - Google example
  - Uses app engine + jsep style protocol
- Test it at https://go.estos.de/chat
  - Uses XMPP (strophe + prosody)
  - Uses Jingle (where possible)
  - No registration required
  - Not localized yet

Why should you care?

- Jingle development started in 2005
  - Slow adoption
  - because XMPP developers do not have enough A/V knowledge?
- webRTC allows XMPP developers to add Audio/Video to their clients (webbased, C++ and Java) with minimal AV knowledge
- Concentrate on signalling / call models
  - „every web developer can now do A/V“?
  - signalling / call model is the hard part
  - many more xmpp developer can now do A/V!

- webRTC makes it easier to build communication islands
  - Because webRTC does not mandate a signalling protocol
  - Establishing XMPP + Jingle + Federation as de-facto standard

URLs

- http://www.youtube.com/watch?v=E8C8ouiXHHk
- http://vimeo.com/52510068
- http://vimeo.com/cullenfluffyjennings/rtcwebexplained

- https://apprtc.appspot.com/
- https://go.estos.de/chat

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webRTC Standards

- Reusing IETF standards (RTC-WEB WG)
  - ICE for NAT traversal
  - DTLS, SRTP, SCTP
  - SDP for negotiating codecs etc
- G.711 and Opus MTI audio codecs
- Video codec…
  - Google and Firefox do VP8
  - The usual debate about VP8 vs H.264/AVC
  - H.264/SVC anyone?
- Javascript API in browsers defined at W3C
- No signalling

A short history of webRTC

- 2005: Google releases libjingle
  - XMPP + media transport library
  - Link your own media lib; gips / mediastreamer (linphone)
- 2008: Google buys on2, releases VP8 video codec in libvpx
- 2010:
  - May: Google buys Global IP Solutions (GIPS)
  - October: RTC-WEB kickoff meeting
- 2011:
  - IETF / W3C working groups formed
  - March: Google releases GIPS engine as webrtc(.org) library
- 2012:
  - January: webRTC supports lands in Chrome
  - October: webRTC enabled by default in chrome M23
  - November: webRTC support in Firefox Nightly

Javascript API

- getUserMedia API
- Peerconnection API
  - createOffer, createAnswer
  - setLocalDescription, setRemoteDescription
  - addIceCandidate, onicecandidate
- SDP „blobs“ (underspecified ones according to MSFT) are used
- JSEP „protocol“, JSON-encoded type + sdp
- „every web developer can now do voip“

Native C++ API

- webrtc.org
- BSD license, C++, cross-platform
- used by Chrome and partially Firefox
- provides media / networking engine
- hook it up with your signalling protocol