WebRTC @ FOMS 2013

WebRTC core

- Codecs
  - Huge fight!
- Protocols
  - SDP
  - ICE (STUN/**TURN**)
  - DTLS-SRTP
  - RTCP-FB
- JS API
  - A lot of async stuff

Different apps, different reqs

- Different apps may require different things
  - e.g. Streaming is recvonly, Conferencing is not
- Is SDP too much, or too less?
  - For AV, maybe often too much
    - But how much can we safely remove?
  - Cover what's missing somehow else?
- How to do media setup?
  - Several libraries use SIP (JsSip, sipml5, etc.)
  - Generic approach (e.g. JSON) better in some cases?

Making things easier for devs

- WebRTC JS API quite complex
  - Everything is async
  - Some sort of library a la jQuery?
    - At different levels, maybe
- No way of reducing it to a URL
  - “Is there any WebRTC Icecast equivalent?”
  - e.g. webrtc://mydomain/myapp/mystream
  - Is this reasonable (or even possible)?