WebRTC

The Web Way to Communicate

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References

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- NVP (Network Voice Protocol) - 1977
- RTP (Real-time Transport Protocol) -1992
  - First published as IETF RFC in 1996
  - Still used today for VoIP and with SIP
- ITU H.323 video telephony standard - 1996
- IETF SIP (Session Initiation Protocol) - 1999
  - Unleashed VoIP revolution on telphony
  - Video and room conferencing
  - Protocol widely used by service providers and in enterprises
- Real-Time Communcation on the Web
  - Voice and video on the Internet using browser plugins
  - 2006 with GoogleTalk inside Gmail
  - WebRTC standardizes and eliminates need for plugin or download

Why not just use Flash?

Pros

- Most browser already have Flash plugin
- Streaming audio and video uses Flash today
- Flash supports real-time audio and video
- Web developers familiar with Flash

Cons

- Flash is single-vendor proprietary and closed
- Losing market share and not available on iOS
- Limited codec and echo cancellation options
- Proprietary development tools

WebRTC is “Skype in the browser”

- Access to camera and microphone without a plugin ($\rightarrow$ No more Flash!)
- Audio/video direct from browser to browser
- Why does it matter?
    - Media can stay local
    - Mobile devices eventually dropping voice channel anyway
    - Games

- New Browser Real-Time Communication (RTC) Function built-in to browsers
- Contains
  - Audio and video codecs
  - Ability to negotiate peer-to-peer connections
  - Echo cancellation, packet loss concealment
- In Chrome and Mozilla today

So What's the Big Deal

- The web is now a platform for real-time communications development
- Communication will be secure (encrypted) by default
- Latest audio and video codecs means superior quality to anything else
- WebRTC provides peer-to-peer media, even through NATs
- Standard that can interoperate with existing VoIP, video conferencing, and even PSTN

- Both browsers running the smae web application from web server
- Peer Connection media session is establised between them
- Signaling is not standardized, could be SIP, Jingle, proprietary. Uses HTTP or WebSockets for transport

WebRTC Trapezoid

- Similar to SIP Trapezoid
- Web Servers communicatie using SIP or Jingle
- Useful for building conventional telephony apps, but unclear how this works in the web world

Questions ?

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