

WebRTC: The Web Way to Communicate

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WebRTC Tutorial Topics

- What is WebRTC?
- How to Use WebRTC – Peer Connection
- WebRTC Peer-to-Peer Media
- WebRTC Protocols and IETF Standards
- WebRTC and the Enterprise
- What's Next?

What is WebRTC?

Real-Time Communication on the Internet

- NVP (Network Voice Protocol)
 - 1977
- RTP (Real-time Transport Protocol)
 - First used in 1992
 - First published as IETF RFC in 1996
 - Still used today for VoIP and with SIP
- ITU H.323 video telephony standard
 - 1996
 - Voice and video conferencing
- IETF SIP – Session Initiation Protocol
 - 1999
 - Unleashed VoIP revolution on telephony
 - Video and room conferencing
 - Protocol widely used by service providers and in enterprises
- Real-Time Communication 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

IIT Voice and Video over Web Project

Demo: IIT Web RTC Conference Project

(1) Signaling API
Resource-based, SDP=>XML/JSON,
subscribe/notify, long-lived connection,
persistent vs transient data, access control

(2) Communication Widgets
Click to call, contact list,
conference object

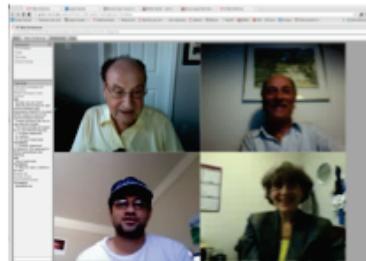
(3) Media Application API
Transport, auth and media objects

Apache web server,
PHP websocket server,
MySQL database

DB

Remote presentation

Video chat



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<http://sites.google.com/site/vvowproject>

Why not just use Flash?

Pros

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

Cons

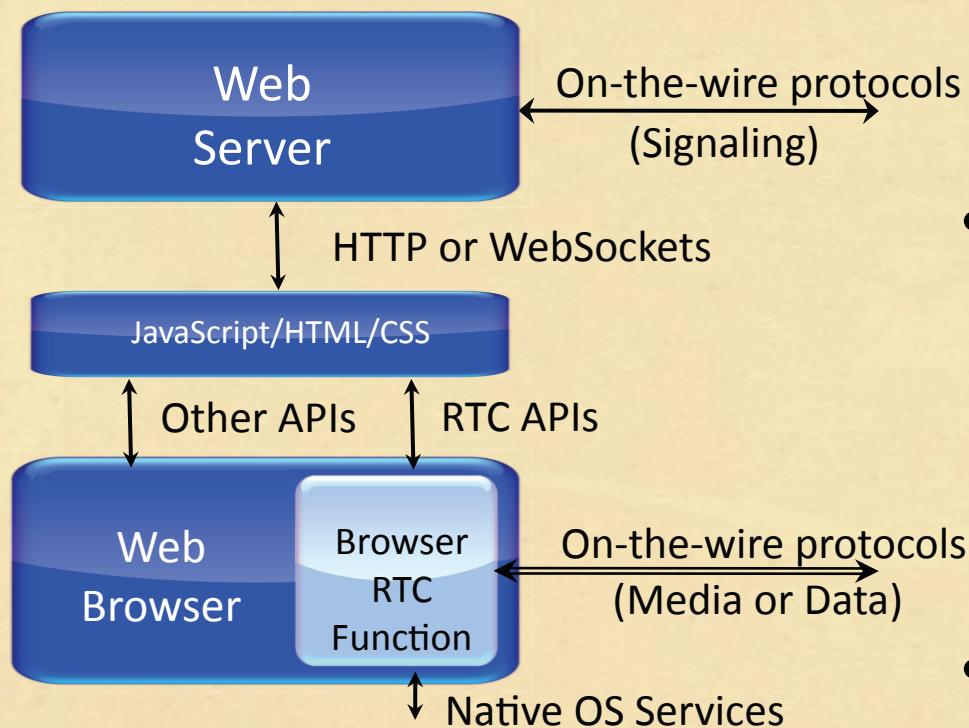
- Flash is single-vendor proprietary and closed
- Losing market share and not available on iOS
- Limited codec and echo cancellation options
- Proprietary development tools



WebRTC is “Skype in the browser”

- Access to camera and microphone without a plugin
 - 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

The Browser RTC Function



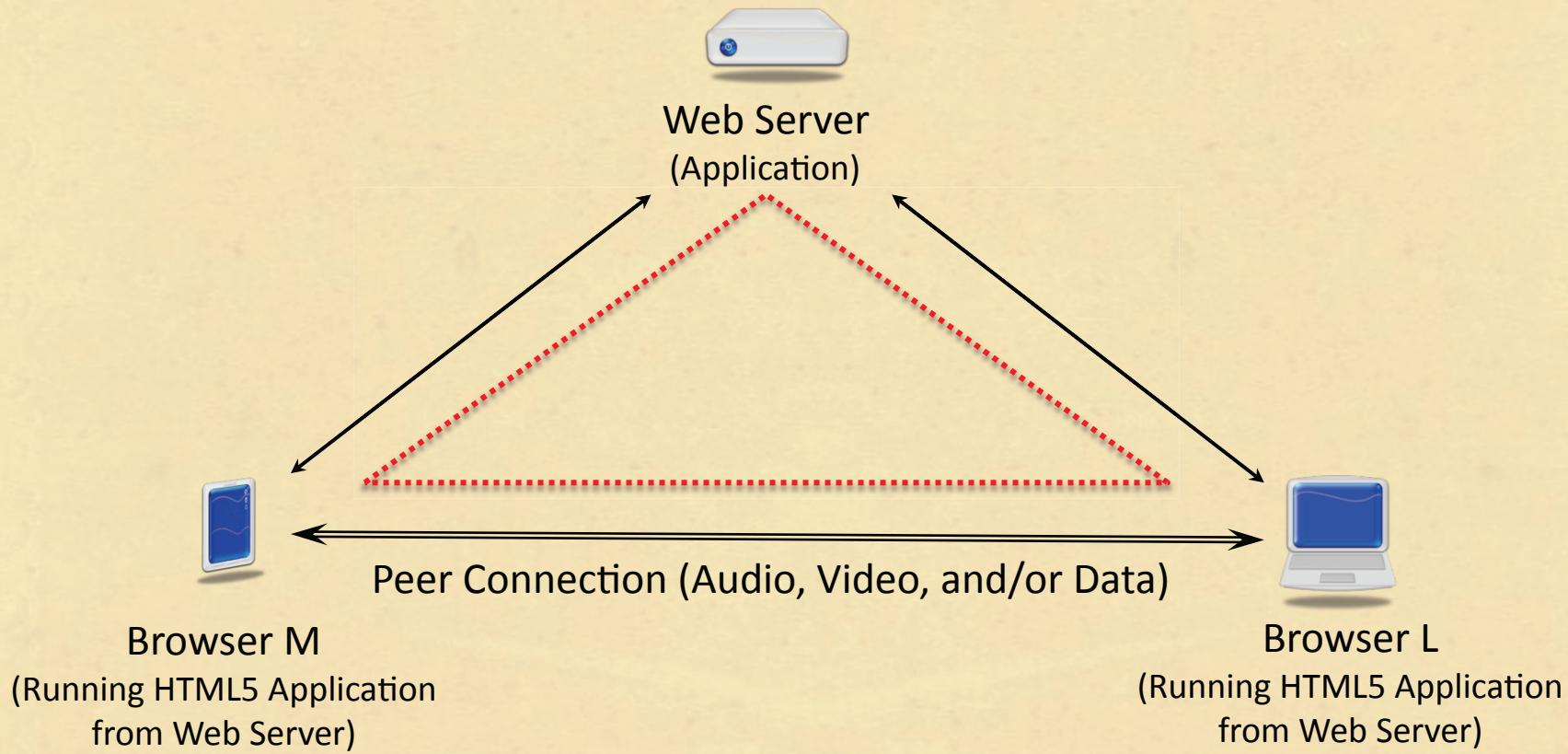
- New Browser Real-Time Communication (RTC) Function built-in to browsers
- Contains
 - Audio and video codecs
 - Ability to negotiate peer-to-peer connections
 - Echo cancellation, packet loss concealment
- In Chrome and Mozilla today, Internet Explorer and Safari eventually

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 (Voice over IP), video conferencing, and even PSTN

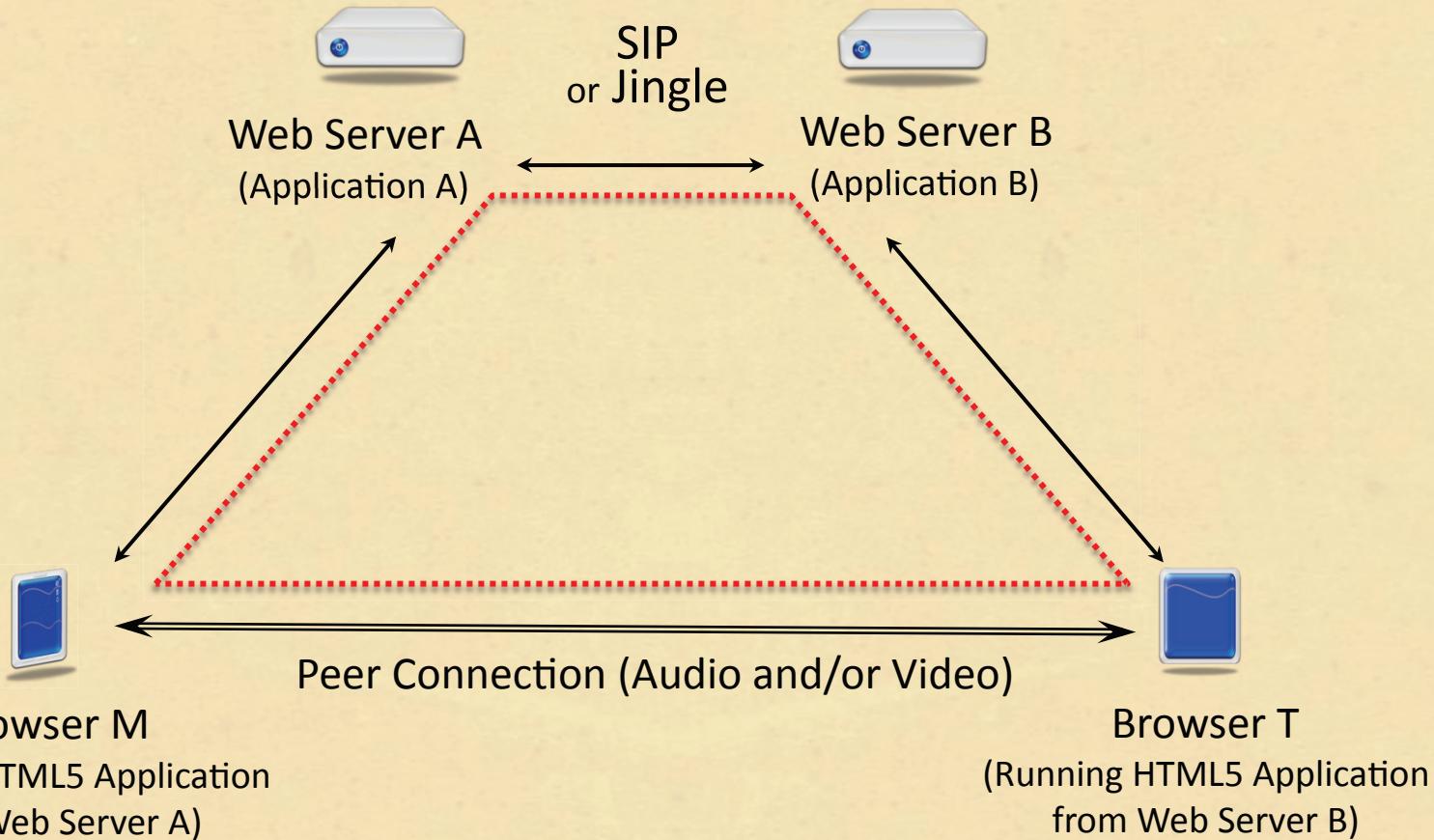
All of these are major innovations for the communications industry

WebRTC Triangle



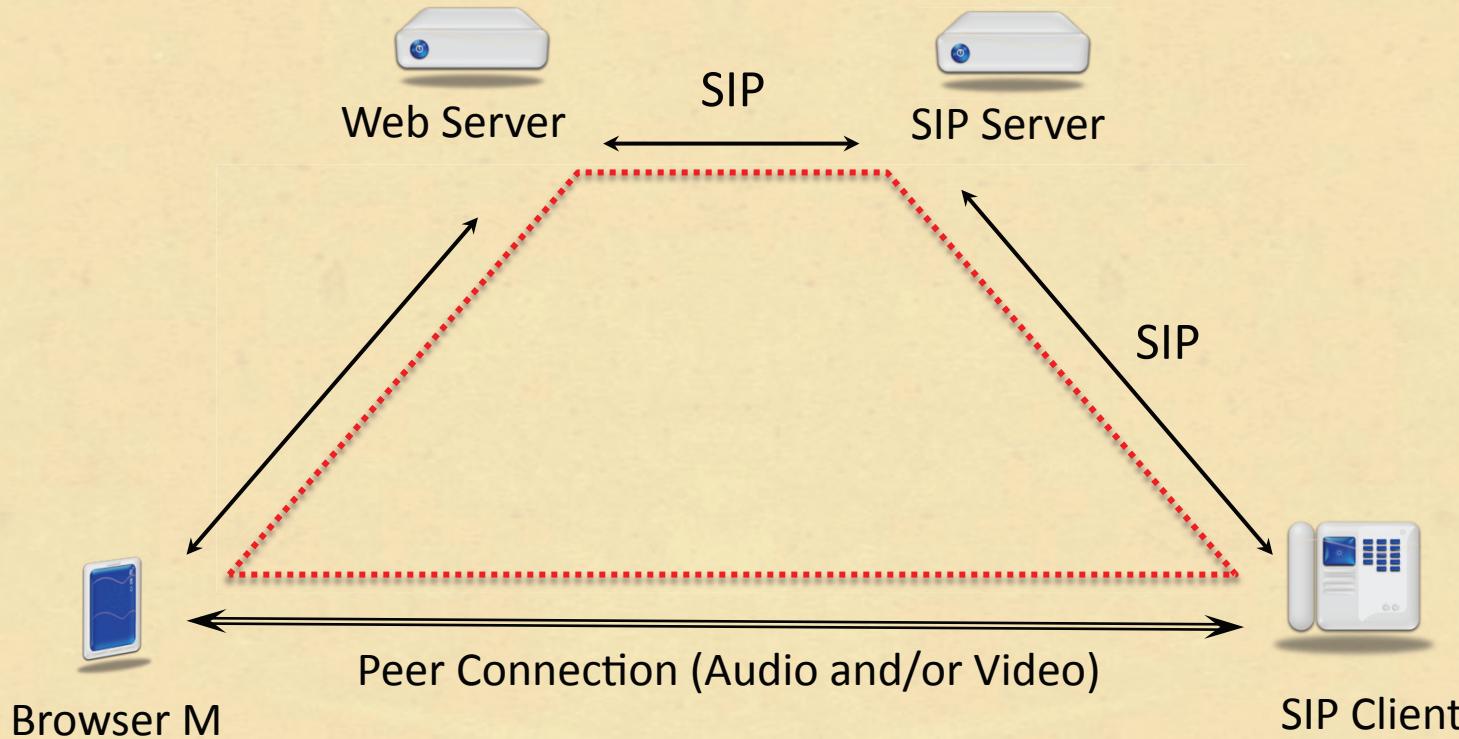
- Both browsers running the same web application from web server
- Peer Connection media session is established between them
- Signaling is not standardized, could be SIP, Jingle, proprietary. Uses HTTP or WebSockets for transport

WebRTC Trapezoid



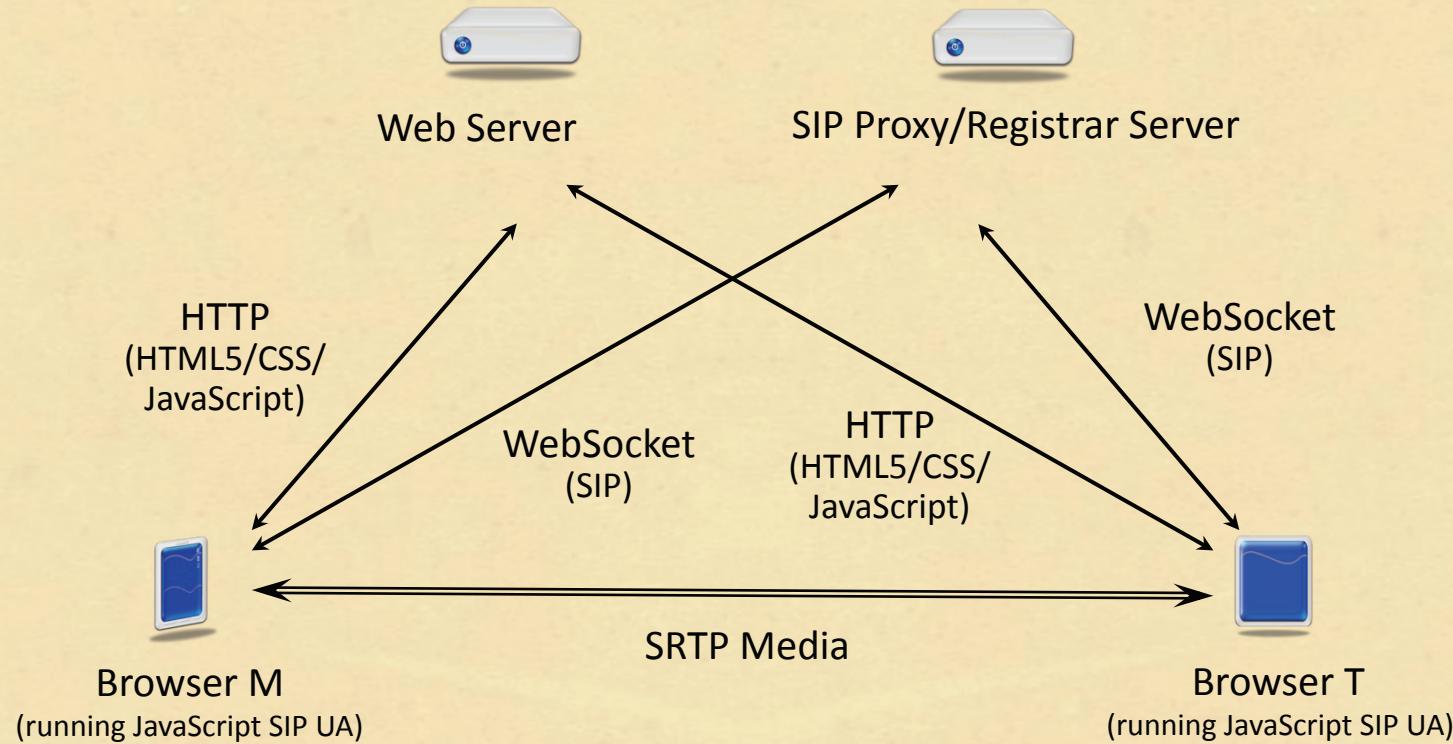
- Similar to SIP Trapezoid
- Web Servers communicate using SIP or Jingle
- Useful for building conventional telephony apps, but unclear how this works in the web world

WebRTC and SIP



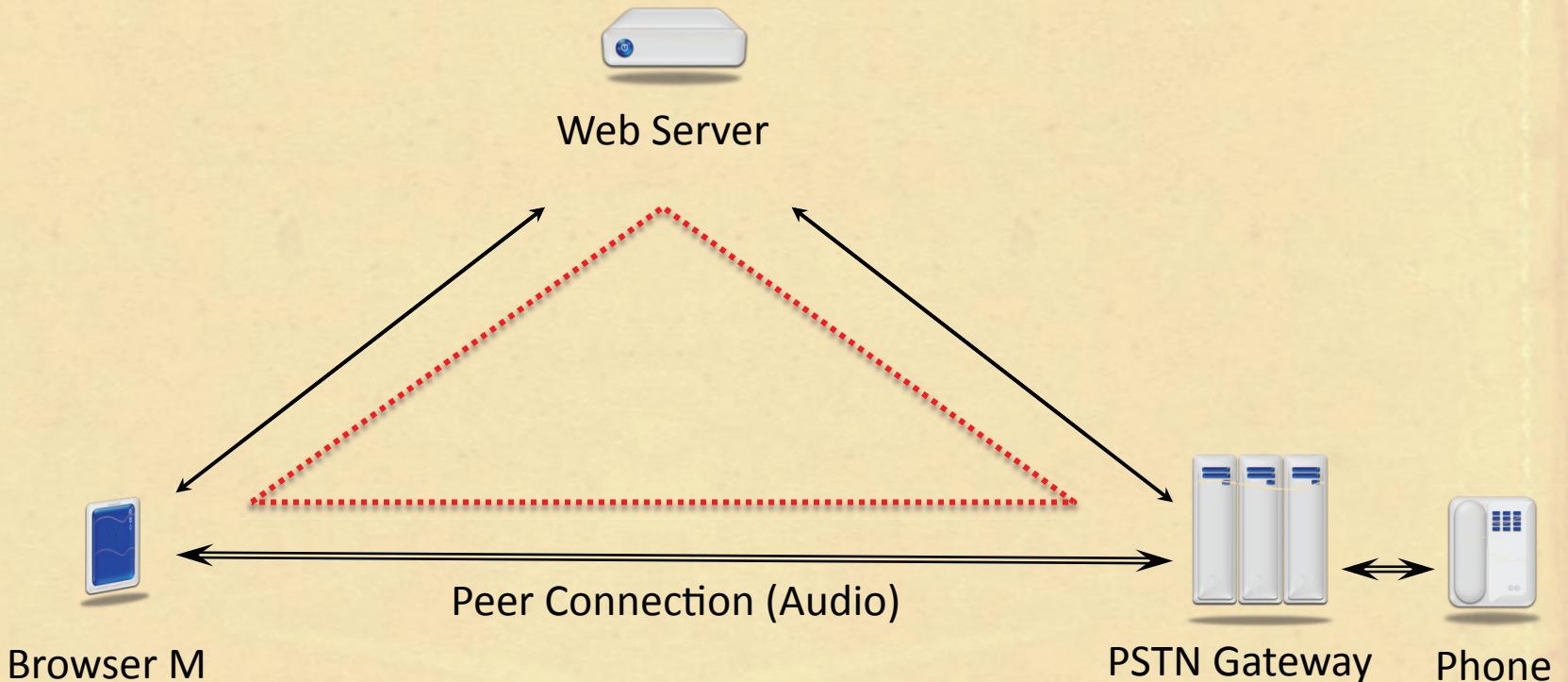
- Peer Connection appears as a standard RTP media session, described by SDP
- Web Server implements a JSEP (JavaScript Session Establishment Protocol) to SIP signaling gateway
- SIP Endpoint must support RTCWEB Media extensions (ICE NAT Traversal, Secure RTP, etc.)

WebRTC with SIP



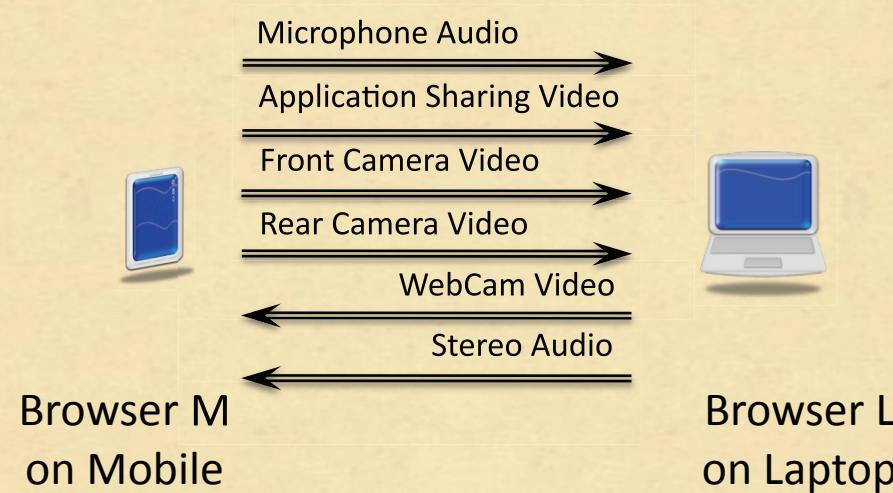
- Browser runs a SIP User Agent by running JavaScript from Web Server
- Secure RTP media connection uses WebRTC APIs
- Details in [draft-ietf-sipcore-websocket] that defines SIP transport over WebSockets

WebRTC and PSTN



- Peer Connection terminates on a PSTN Gateway
- Audio Only
- Could also use SIP trunking such as SIPconnect 1.1 recommendation

WebRTC Support of Multiple Media

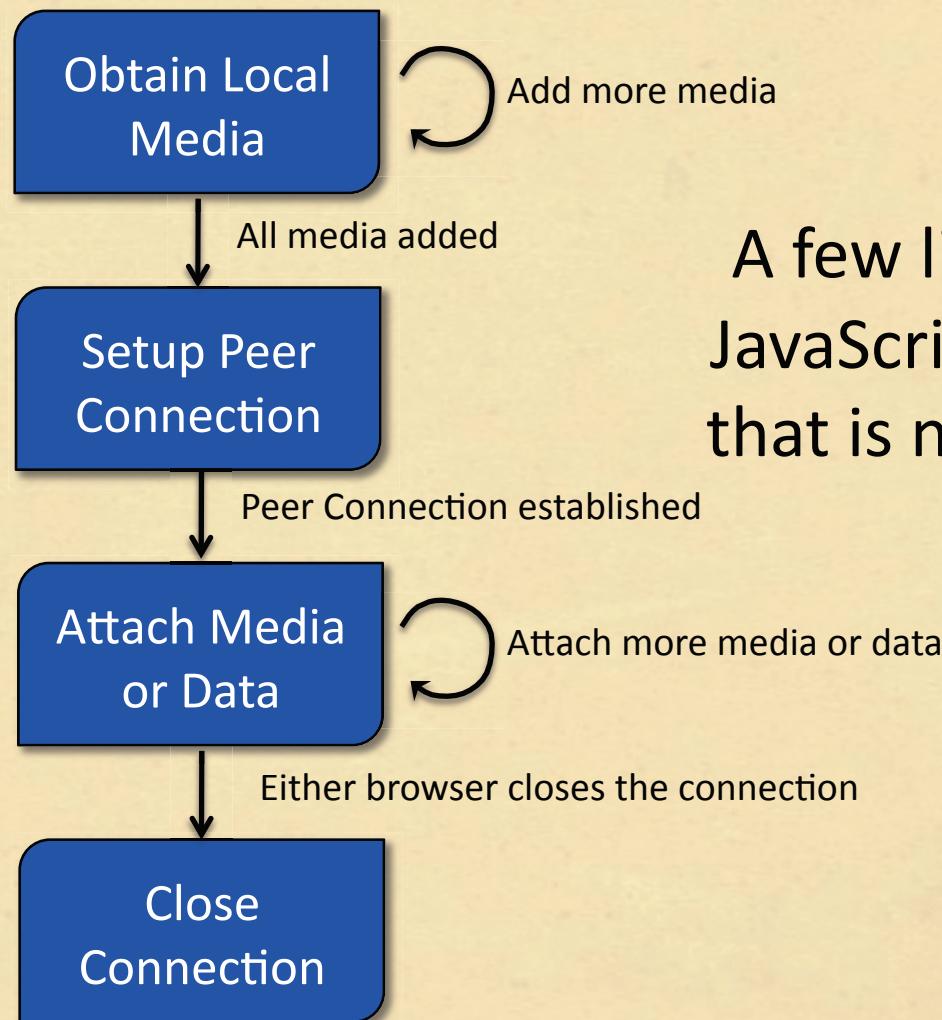


- Multiple sources of audio and video are assumed and supported
- All RTP media, voice and video, and RTCP feedback messages are multiplexed over the same UDP port and address

How to Use WebRTC

How to use WebRTC

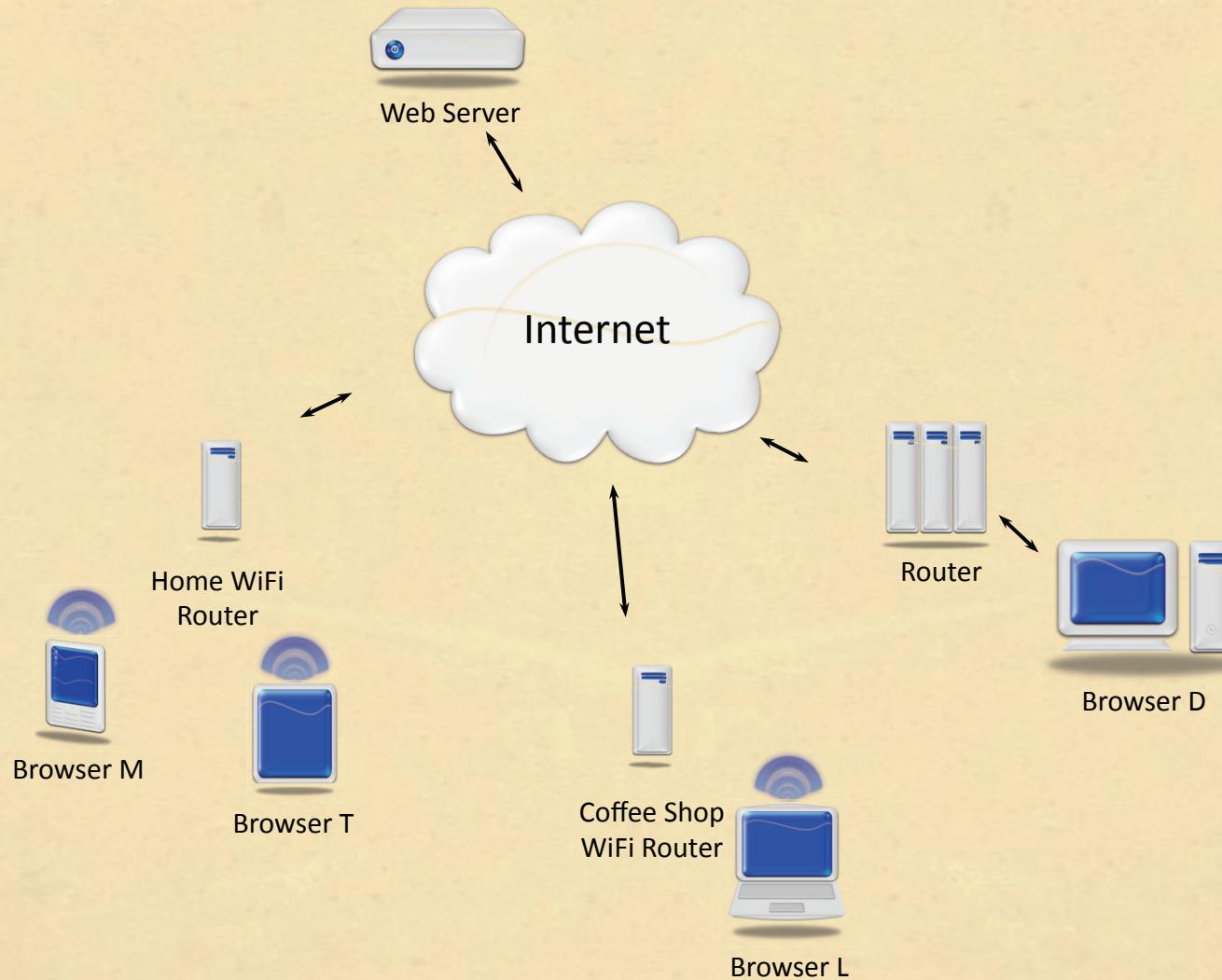
```
getUserMedia ()  
  
RTCPeerConnection ()  
  
addStream ()  
createOffer ()  
createAnswer ()
```



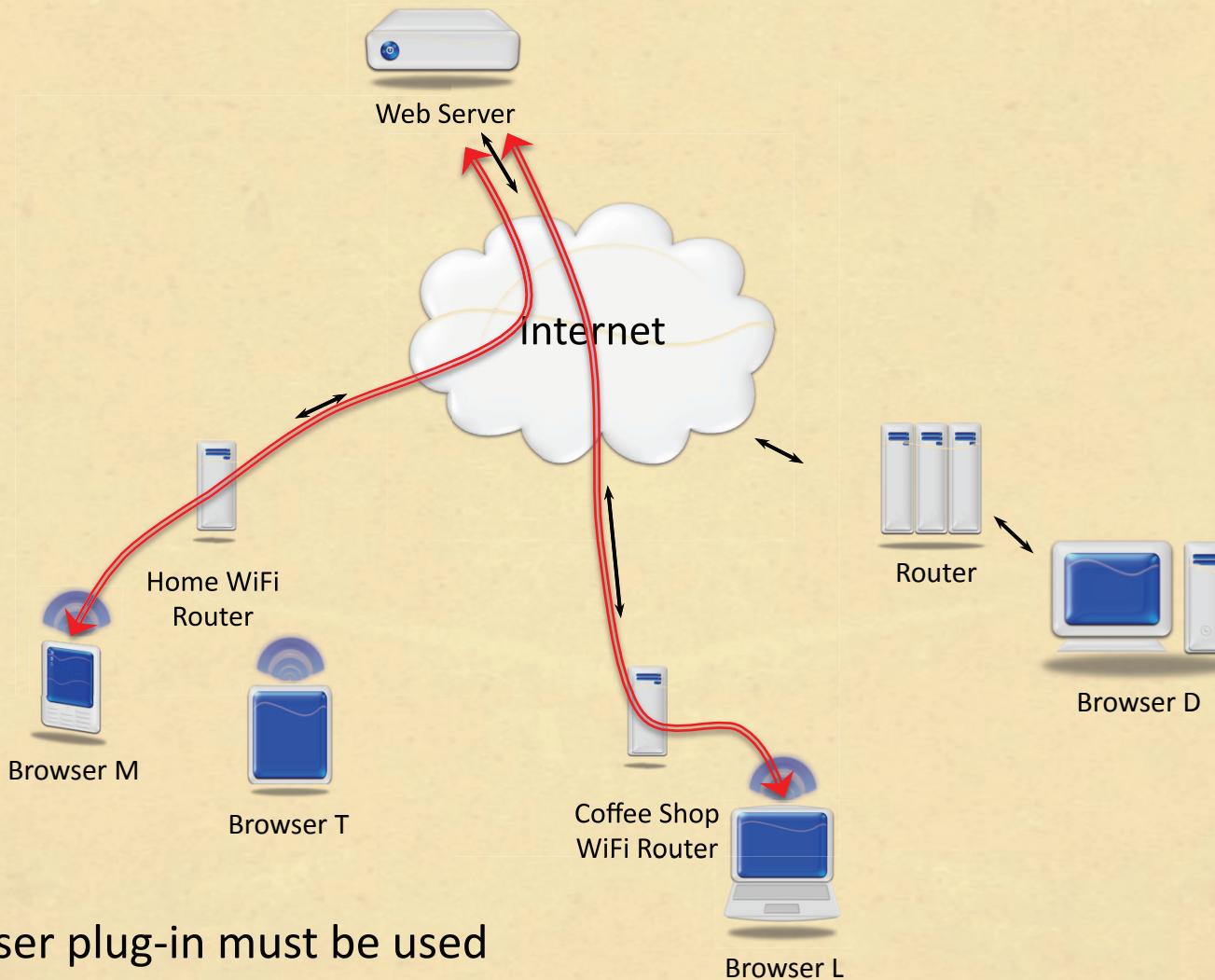
A few lines of
JavaScript is all
that is needed!

WebRTC Peer-to-Peer Media

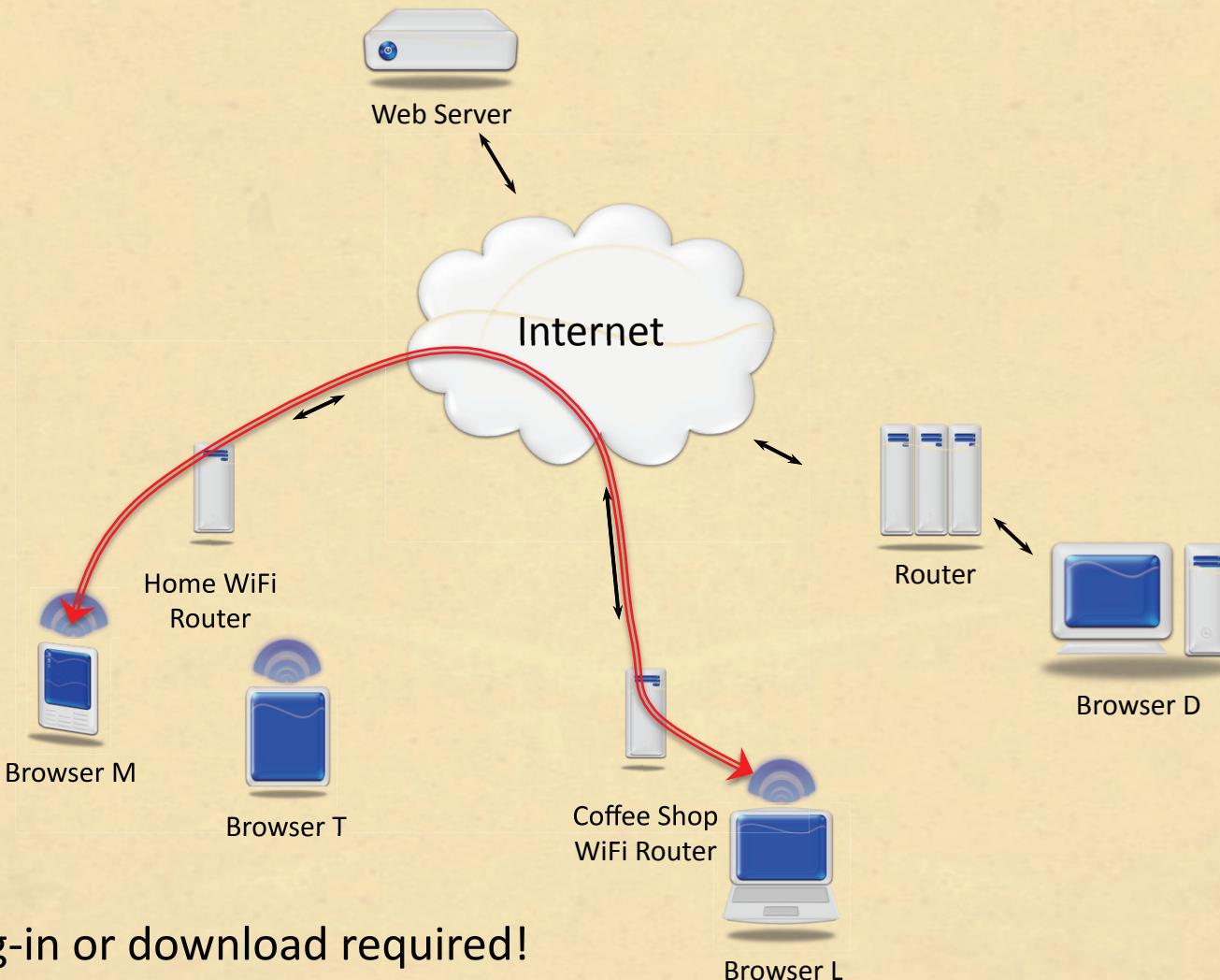
Media Flows in WebRTC



Media without WebRTC

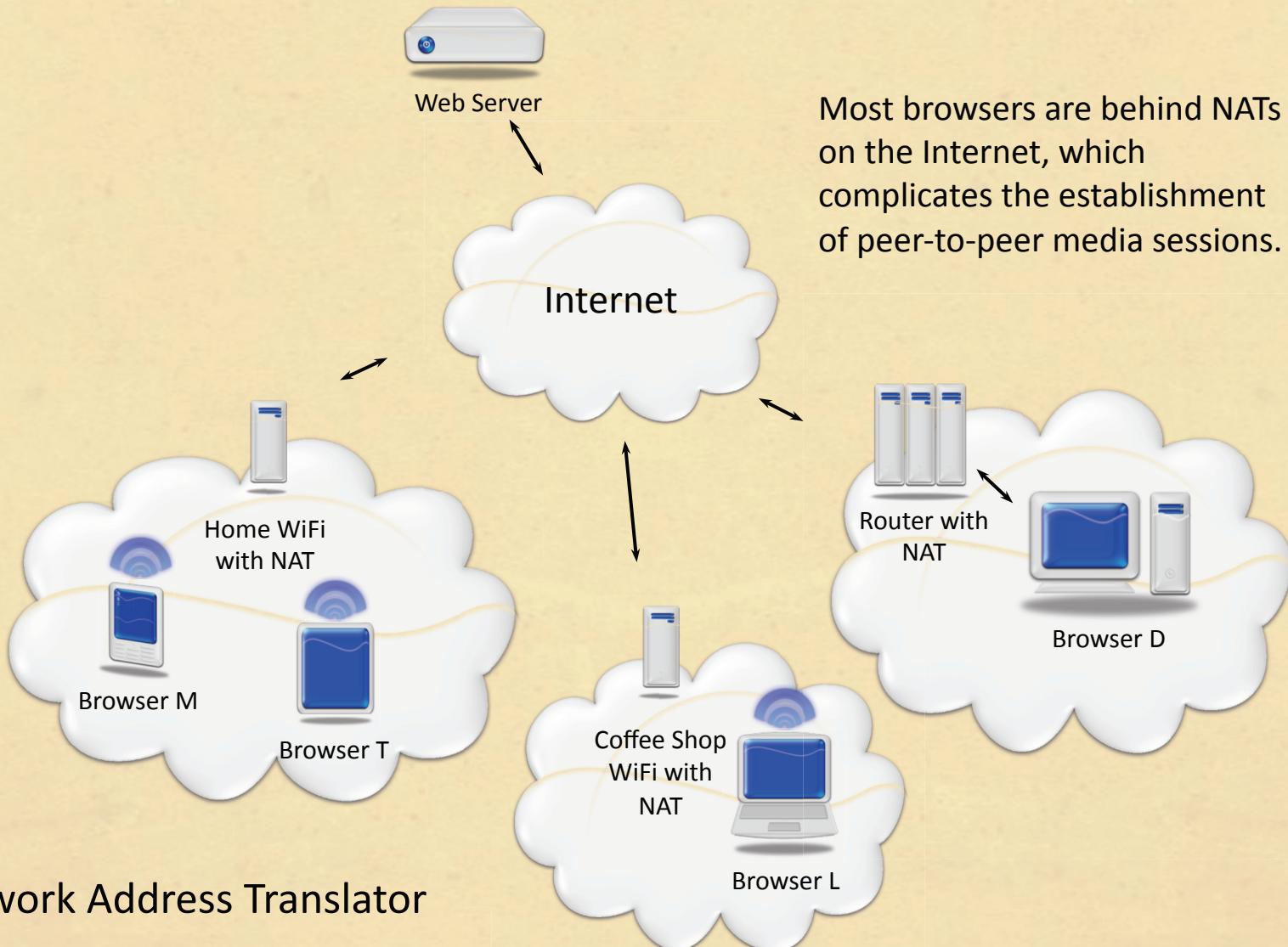


Peer-to-Peer Media with WebRTC

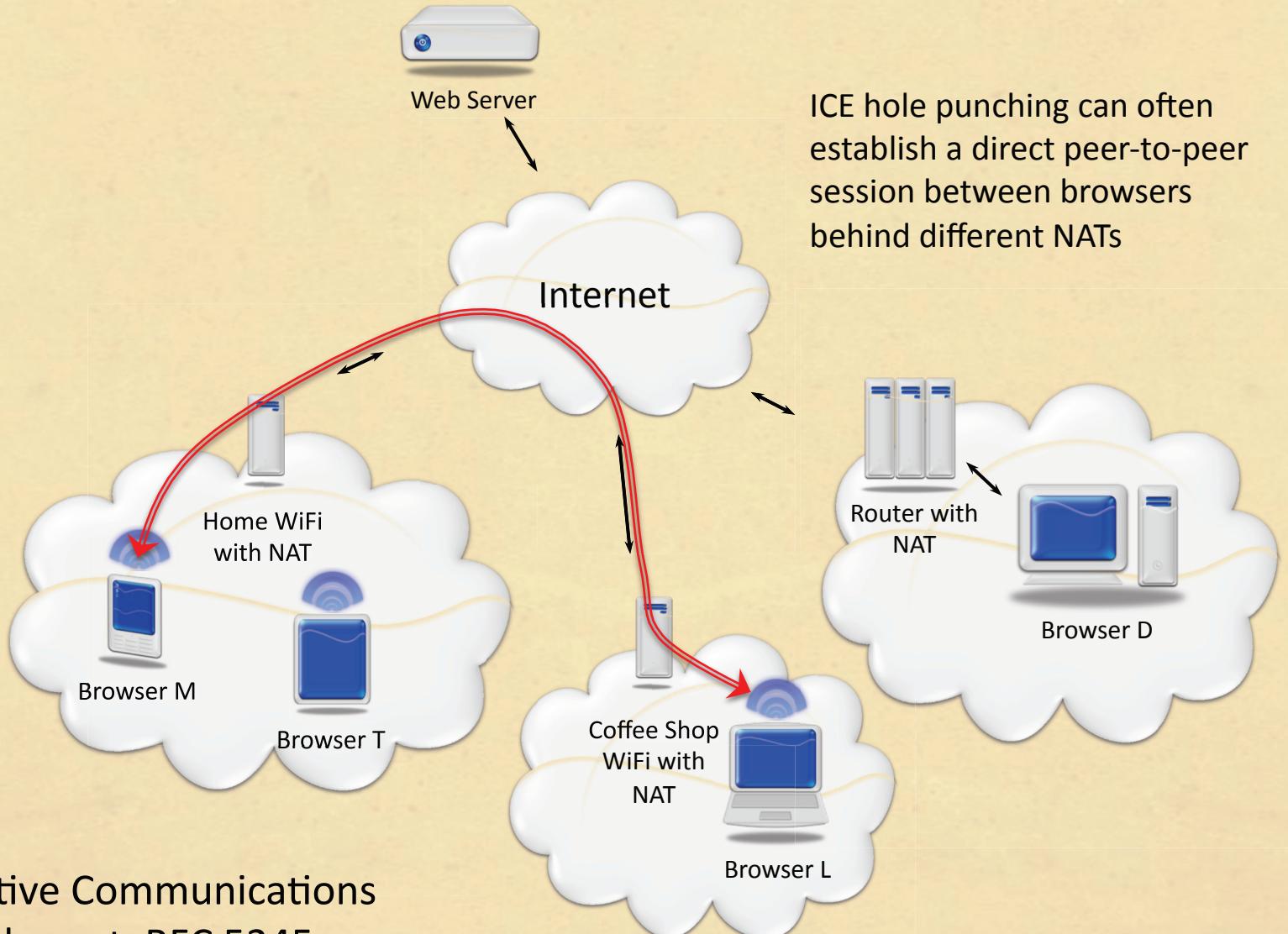


No plug-in or download required!

NAT Complicates Peer-to-Peer Media

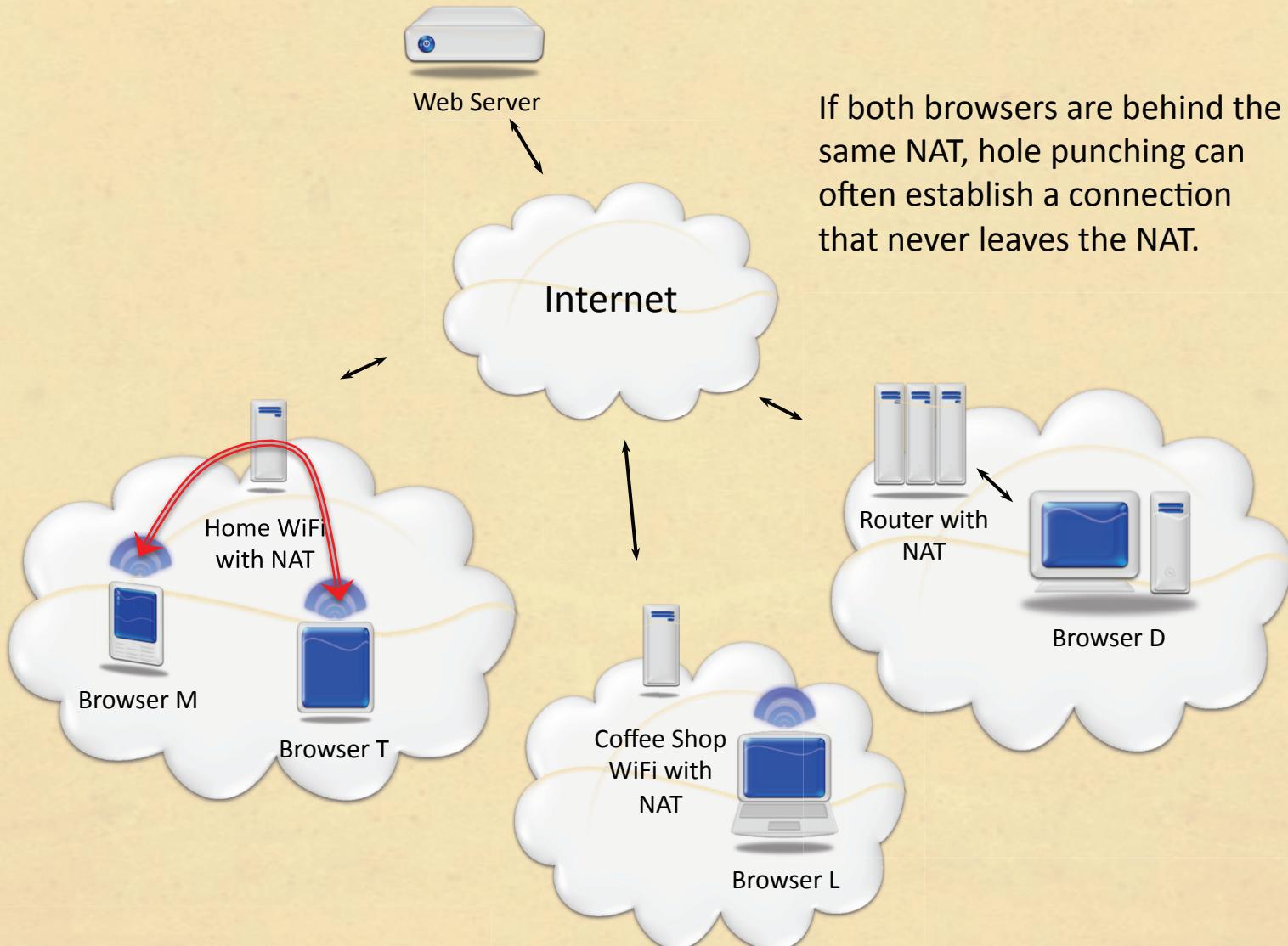


Peer-to-Peer Media Through NAT

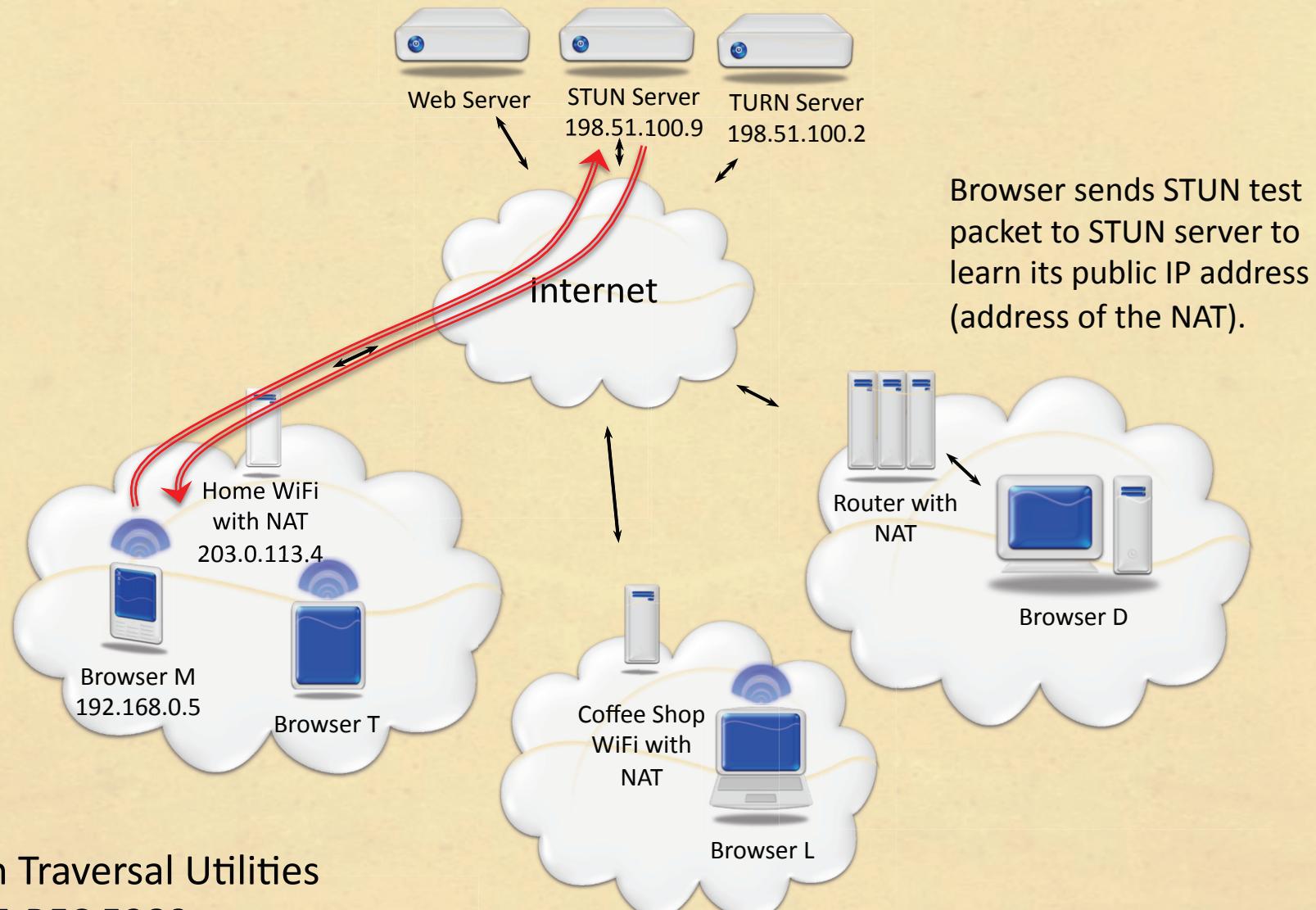


Interactive Communications
Establishment, RFC 5245

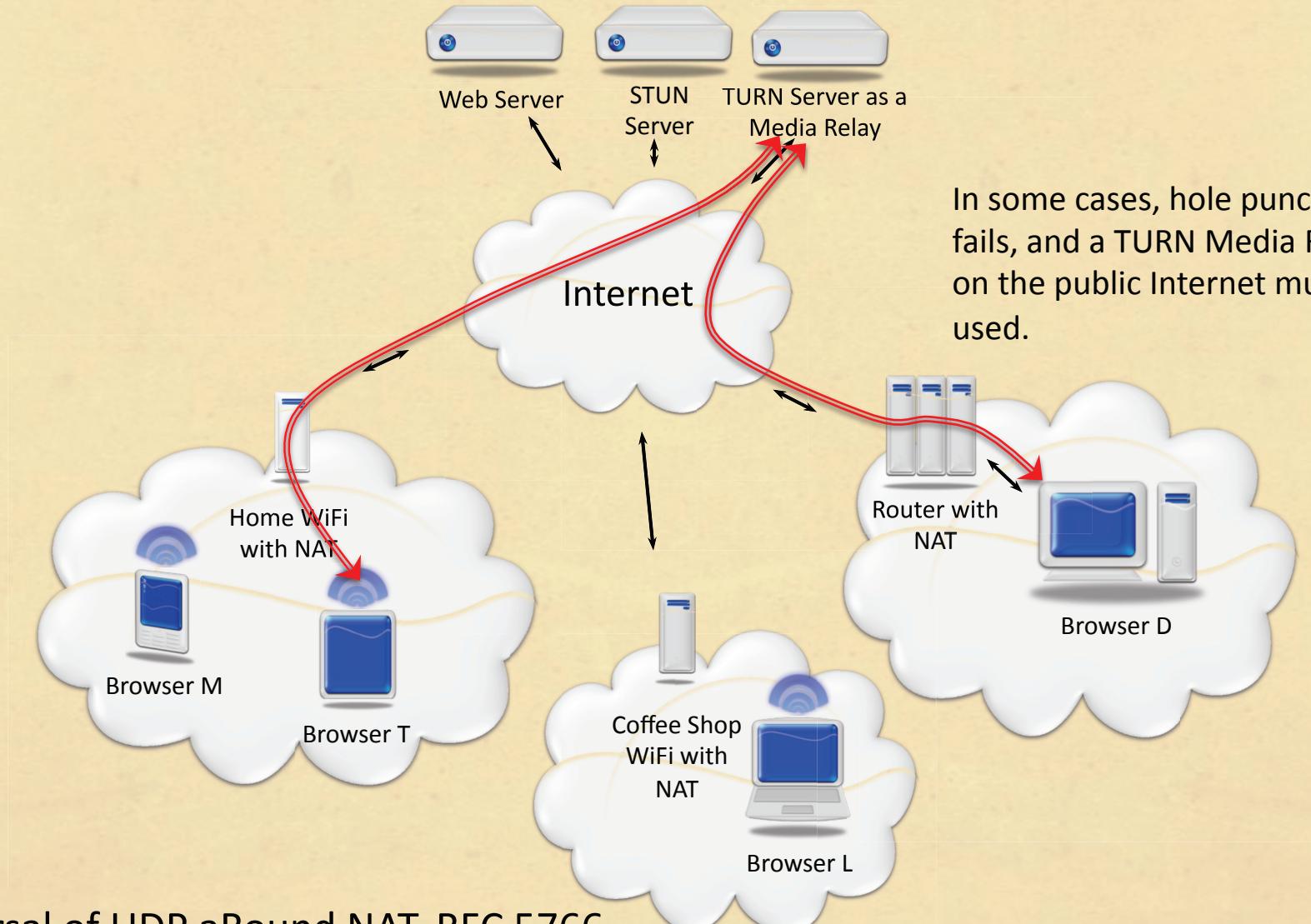
P2P Media Can Stay Local to NAT



Browser Queries STUN Server



TURN Server Can Relay Media



Traversal of UDP aRound NAT, RFC 5766

WebRTC Protocols and IETF Standards

WebRTC Protocols

Application Layer



Transport Layer



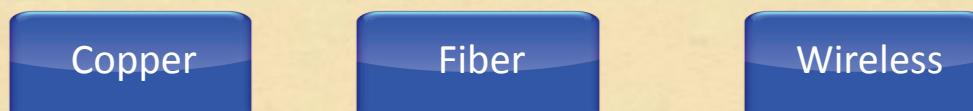
Network Layer



DataLink Layer



Physical Layer



A Joint Standards Effort

- World Wide Web Consortium (W3C)
 - Standardizing APIs (Application Programming Interfaces)
 - Most work in WEBRTC Working Group
 - Used by JavaScript to access RTC function
- Internet Engineering Task Force (IETF)
 - Standardizing protocols (bits on the wire)
 - Codecs (more on this next)
 - Peer Connection will use RTP, SDP, and extensions
 - Some work in RTCWEB Working Group
 - Lots of related work in MMUSIC, AVTCORE, etc.



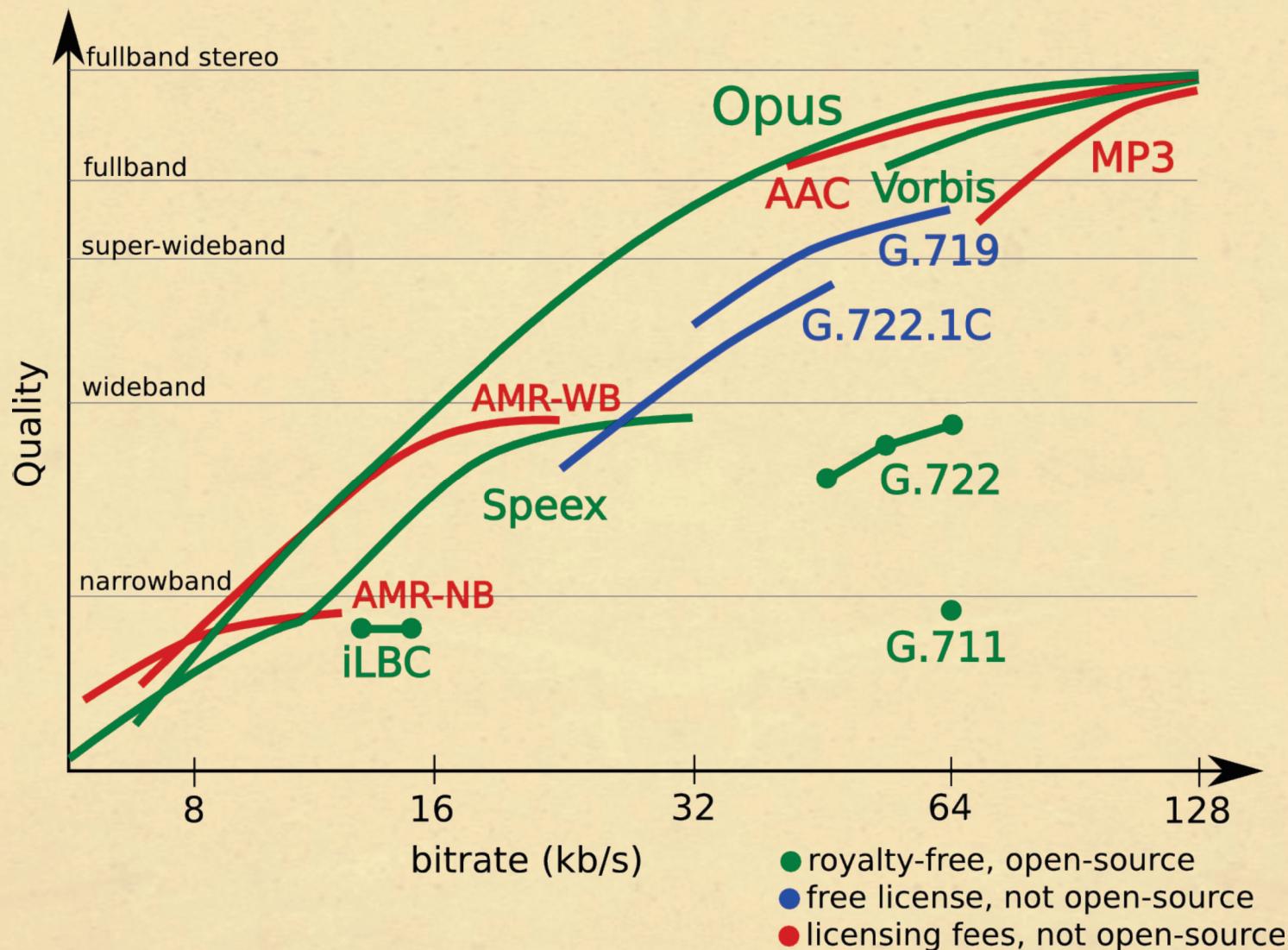
Opus Audio Codec

Codec Feature Comparison

Codec	Sample rate (kHz)	Bitrate (kbps)	Frame size (ms)	Total delay (ms)	Robustness	License
Opus	8-48	6 - 255 (mono) 12 - 510 (stereo)	2.5 - 20	5 - 22.5	packet loss, limited bit errors	Open-source (BSD)
Speex	8 16 32	2.2 - 24.6 4 - 42.2 4.2 - 44	20	30 - 35	packet loss	Open-source (BSD)
G.722.1C (Siren14)	32	24, 32, 48	20	40	packet loss, bit errors	no charge, but not open-source
iLBC	8	15 13.3	20 30	25 40	packet loss	Open-source (BSD)
AAC-LD	16 - 48	16 - 128	10 - 11.6	20 - 50+	packet loss	proprietary, MPEG

From <http://opus-codec.org/comparison/>

Audio Codec Comparison



<http://opus-codec.org/comparison/quality.png>

Standard Codecs in WebRTC

Codec	Use	Specification
Opus	Narrowband to wideband Internet audio codec for speech and music	RFC 6716 .
G.711	PCM audio encoding for PSTN interworking and backwards compatibility with VoIP systems	RFC 3551
Telephone Events	Transport of Dual Tone Multi Frequency (DTMF) tones	RFC 4733
H.264	Video codec requiring licensing	RFC 6184
VP8	Open source video codec	RFC 6386

- Mandatory to Implement (MTI) audio codecs are settled on Opus and G.711 (finally!)
- Video is not yet settled
 - H.264 vs VP8 fight is ugly

WebRTC and the Enterprise

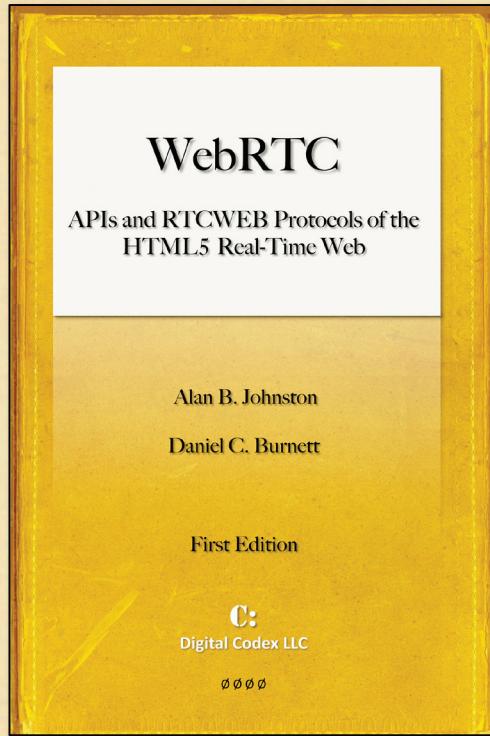
- Enterprise has unique requirements on WebRTC
- Security
 - Firewall traversal
 - Access control
 - Peer-to-peer data flows
- Compliance
 - Recording & logging
 - Policy compliance
- Integration with existing infrastructure
- New element proposed:
 - “Secure Edge” located in enterprise DMZ

See our article “Taking WebRTC to the Enterprise” in April IEEE Communications Magazine

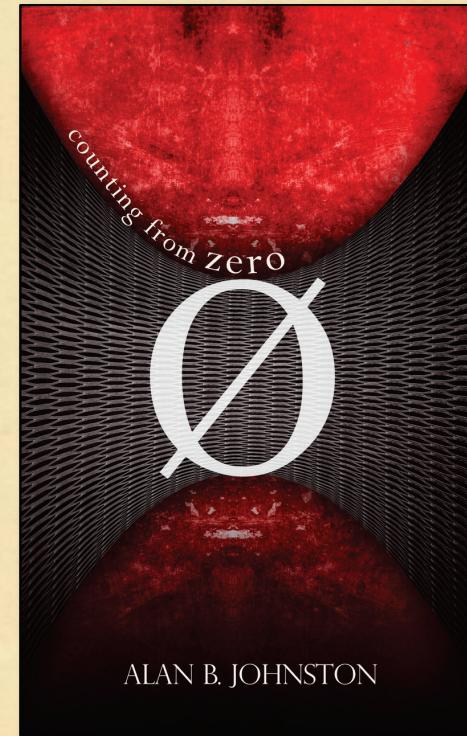
What's Next?

- W3C and IETF standards still need to be finalized (early 2014)
- Browsers need to add support
 - Chrome browser has much of this functionality now!
 - Firefox will have shortly (in nightly builds)
 - Mobile browsers need to support
 - In Android Beta now
- Mandatory to Implement video codec needs to be decided
- Enterprise use of WebRTC need to be worked out

Questions?



<https://webrtcbook.com>



<https://countingfromzero.net>

References

1. C. Davids, A. Johnston, K. Singh, H. Sinnreich, W. Wimmreuter, "SIP APIs for Voice and Video Communications on the Web", Principles, Systems and Applications of IP Telecommunications (IPTcomm), Chicago, IL, Aug 2011
2. A. Johnston, D. Burnett, "WebRTC: APIs and RTCWEB Protocols of the HTML5 Real-Time Web", Digital Codex, St. Louis, MO, 2012
3. A. Johnston, J. Yoakum, K. Singh, "Taking on WebRTC in an Enterprise, (to appear in) IEEE Communications Magazine, Vol 51, No. 4, April 2013