WebRTC

The Web Way to Communicate

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[1] A. B. Johnston and D. C. Burnett, *WebRTC: APIs and RTCWEB Protocols of the HTML5 Real-Time Web*. Digital Codex LLC, 2013. [Online]. Available: http://www.amazon.com/WebRTC-RTCWEB-Protocols-HTML5-Real-Time/dp/098597883X%3FSubscriptionId%3D0JYN1NVW651KCA56C102%26tag%3Dtechkie-20%26linkCode%3Dxm2%26camp%3D2025%26creative%3D165953%26creativeASIN%3D098597883X

[2] R. Manson, *Getting Started with WebRTC*. Packt Publishing, 2013. [Online]. Available: http://www.amazon.com/Getting-Started-WebRTC-Rob-Manson/dp/1782166300%3FSubscriptionId%3D0JYN1NVW651KCA56C102%26tag%3Dtechkie-20%26linkCode%3Dxm2%26camp%3D2025%26creative%3D165953%26creativeASIN%3D1782166300

- 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

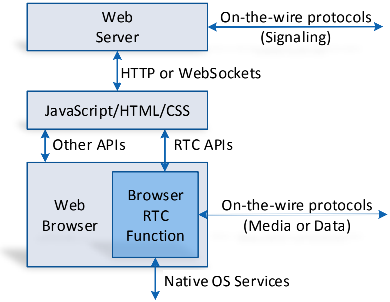
- Flash is single-vendor proprietary and closed
- Losing market share and not available on iOS
- Limited codec and echo cancellation options
- Proprietary development tools

WebRTC is "Skype in the browser"

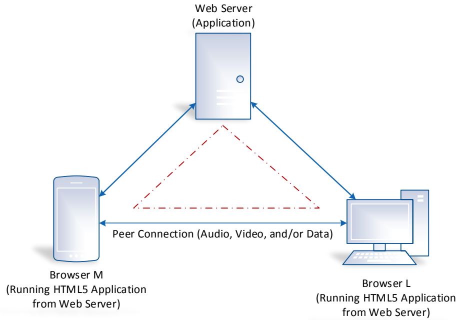
- Access to camera and microphone without a plugin ($\rightarrow$ No more Flash!)
- Audio/video direct from browser to browser
- Why does it matter?
  - Media can stay local
  - Mobile devices eventually dropping voice channel anyway
  - Games

The Browser RTC Function

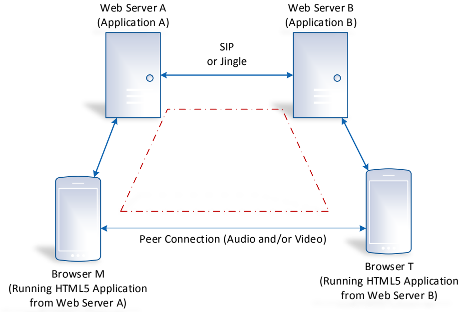
- New Browser Real-Time Communication (RTC) Function built-in to browsers
- Contains
  - Audio and video codecs
  - Ability to negotiate peer-to-peer connections
  - Echo cancellation, packet loss concealment
- In Chrome and Mozilla today



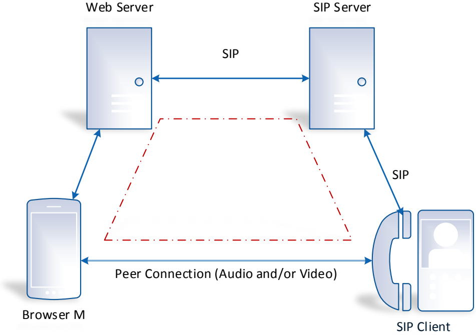
- The web is now a platform for real-time communications development
- Communication will be secure (encrypted) by default
- Latest audio and video codecs means superior quality to anything else
- WebRTC provides peer-to-peer media, even through NATs
- Standard that can interoperate with existing VoIP,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



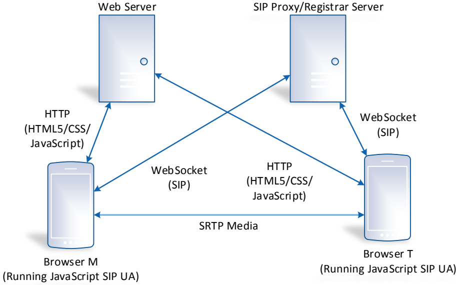
- Both browsers running the smae web application from web server
- Peer Connection media session is establised between them
- Signaling is not standardized, could be SIP, Jingle, proprietary. Uses HTTP or WebSockets for transport



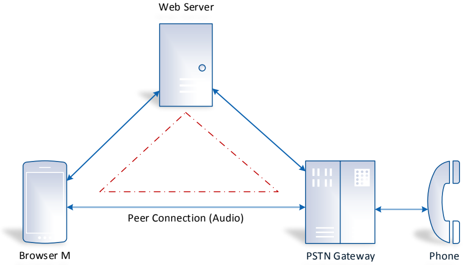
- Similar to SIP Trapezoid
- Web Servers communicatie using SIP or Jingle
- Useful for building conventional telephony apps, but unclear how this works in the web world



- 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



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



- Peer Connection terminates ona PSTN Gateway
- Audio only
- Could also use SIP trunking such as SIPconnect 1.1 recommendation

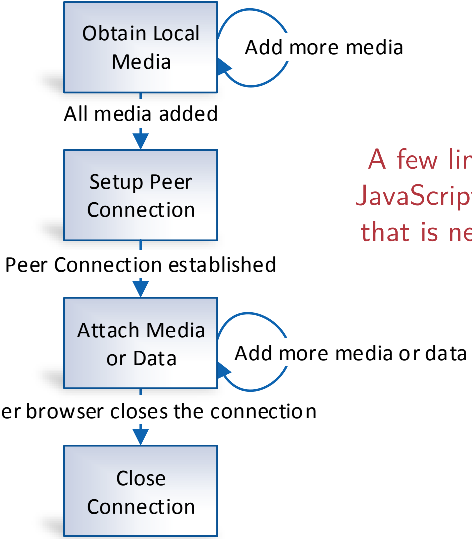
- Multiple sources of audio and video are assumed and supported
- All RTP media, voice and video, and RTCP feedback messages are multiplexed over the same UDP port and address

How to use WebRTC

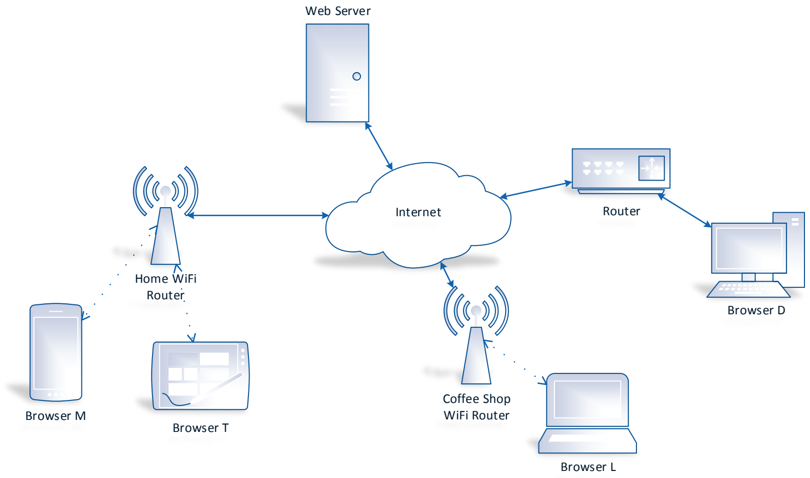
getUserMedia()

RTCPeerConnection()

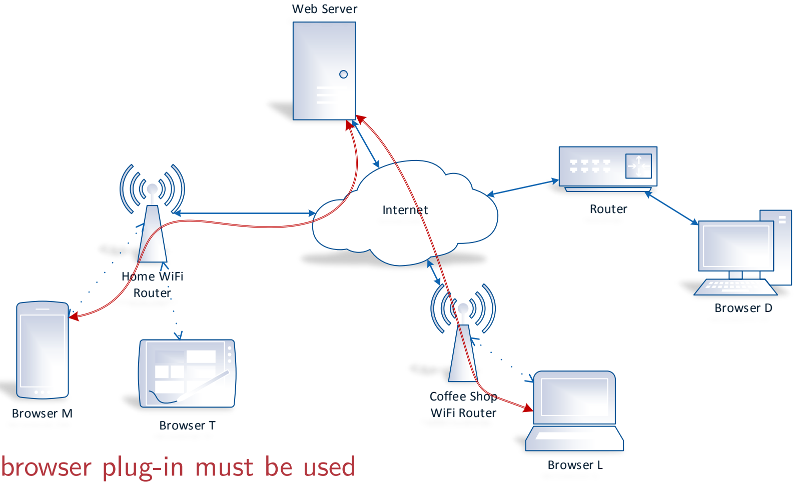
addStream()
createOffer()
createAnswer()



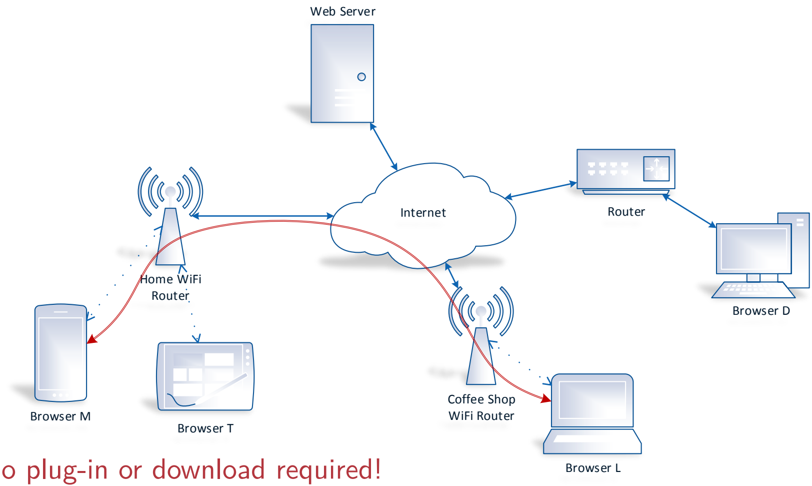
A few lines of JavaScript is all that is needed!



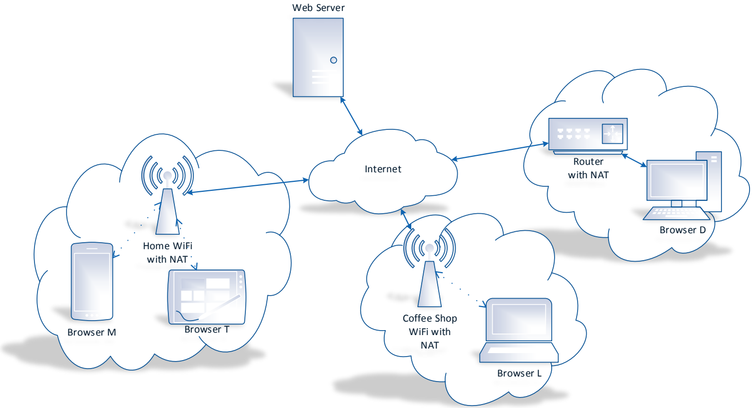
Web Server
Internet
Router
Home WiFi Router
Browser D
Browser M
Browser T
Coffee Shop WiFi Router
Browser L



A browser plug-in must be used

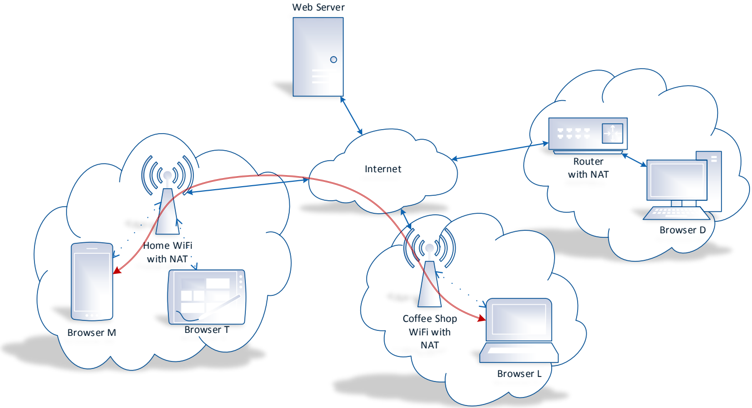


No plug-in or download required!



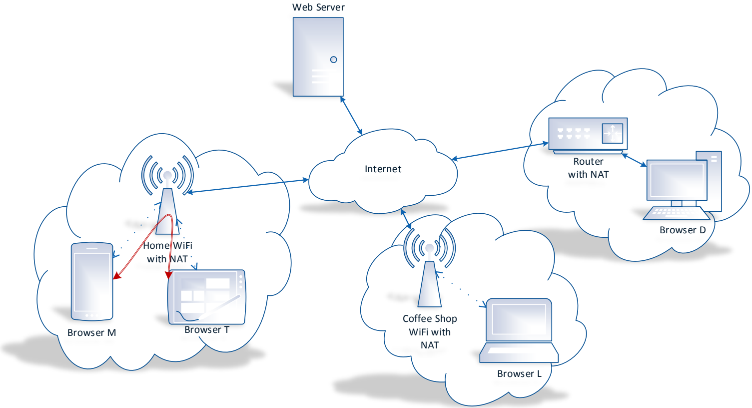
Network Address Translator

Most browsers are behind NATs on the Internet, which complicates the establishment of peer-to-peer media sessions

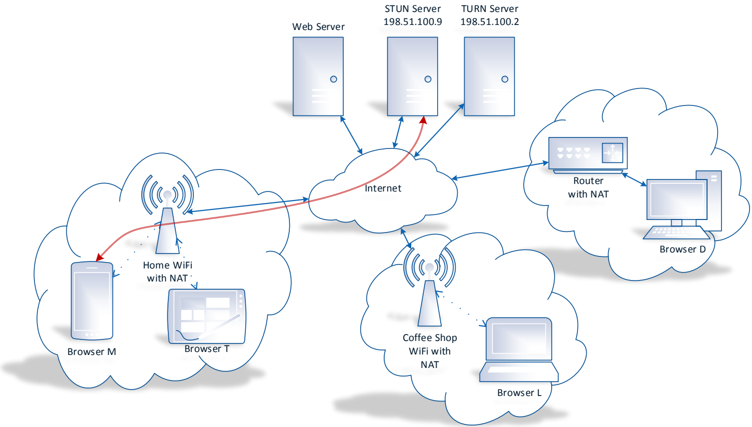


Interactive Communications Establishment, RFC 5245

ICE hole punching can often establish a direct peer-to-peer session between browsers behind different NATs

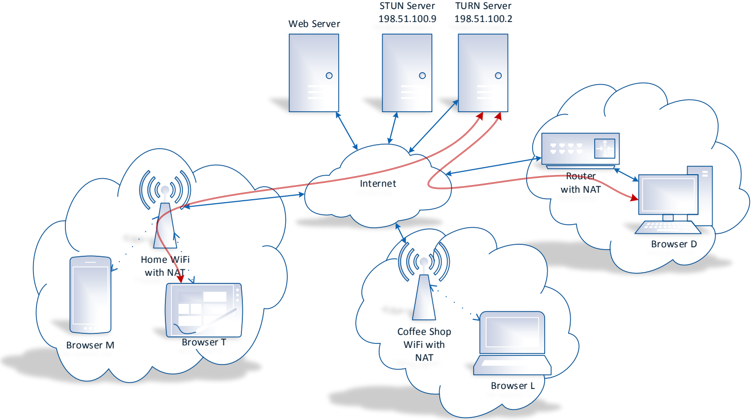


If both browsers are behind the same NAT, hole punching can often establish a connection that never leaves the NAT



Session Traversal Utilities for NAT, RFC 5389

Browser sends STUN test packet to STUN server to learn its public IP address (address of the NAT)



Traversal of UDP aRound NAT, RFC 5766

In some cases, hole punching fails, and a TURN Media Relay on the public Internet must be used.

Questions ?

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