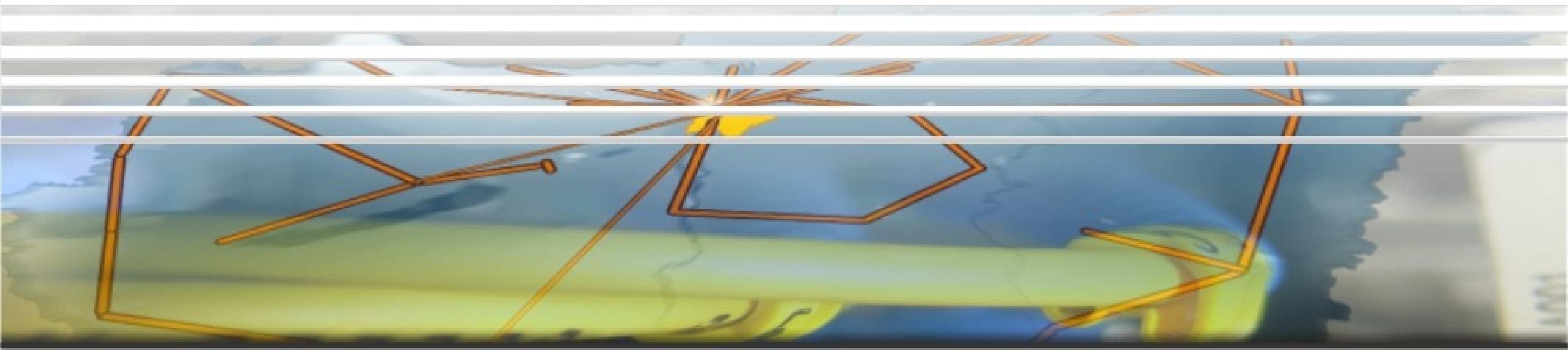
WebRTC

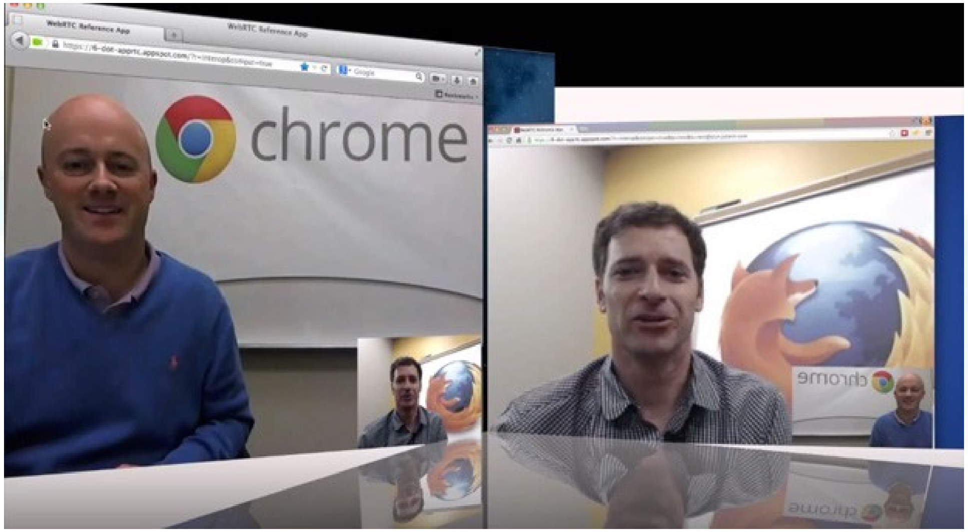


November 7, 2013
Balatongyörök / Hungary

Mészáros Mihály

WebRTC

- WebRTC: “A framework, protocols and application programming interface that provide real time interactive voice, video and data in web browsers and other applications”
- Standardization
  - WEBRTC (W3C) part of HTML5
  - RTCWEB (IETF)
  - / IMS_WebRTC(3GPP) /
- Implementation
  - Chrome
  - FireFox
  - Opera(GetUserMedia only)
  - Browser (Ericsson Research)
- WebRTC native C++ API support (for Browsers and Apps)
  - Android, iOS(coming..)



WebRTC

Javascript Session Establishment Protocol (JSEP)
IETF RTCWEB WorkGroup

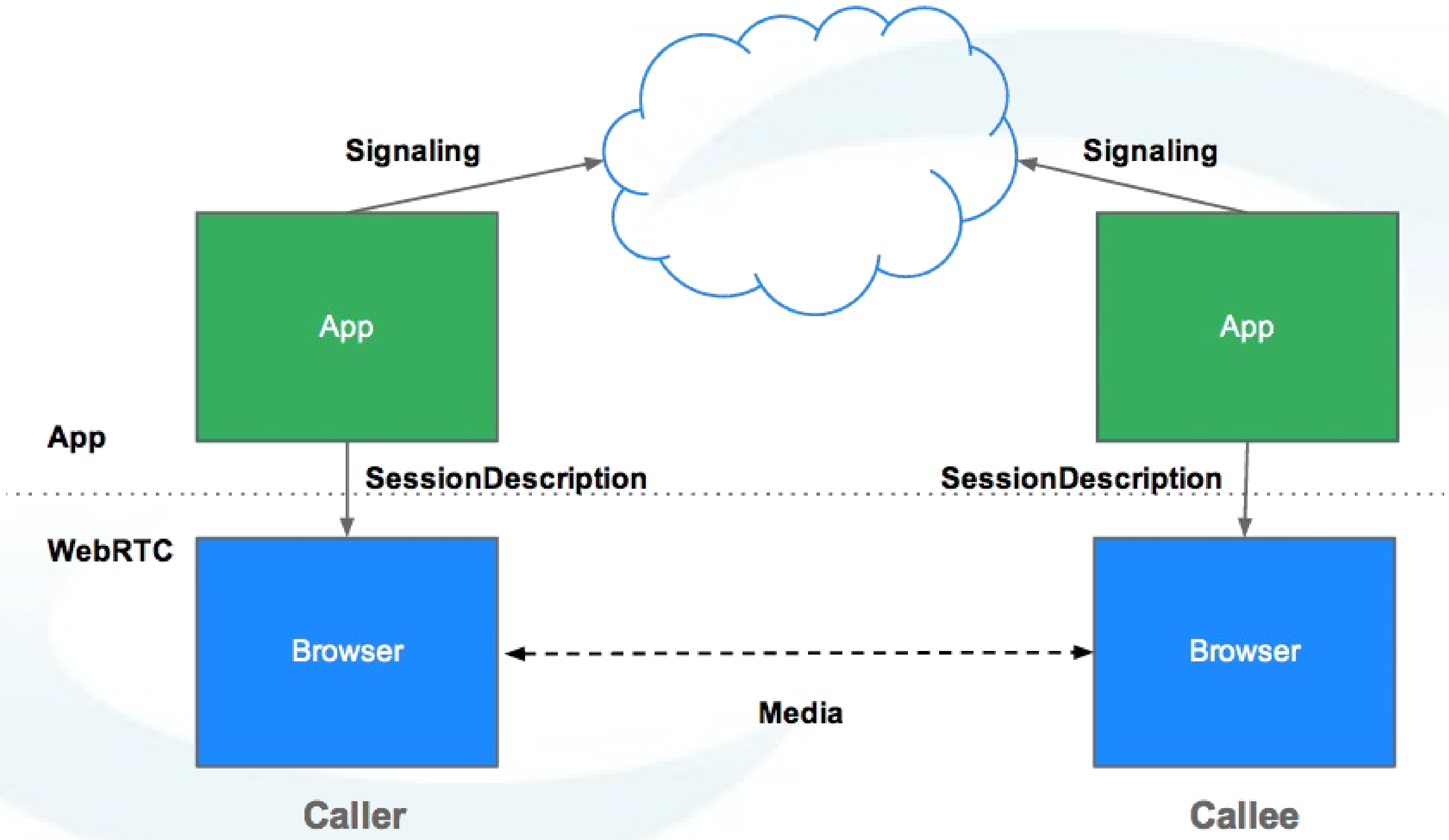
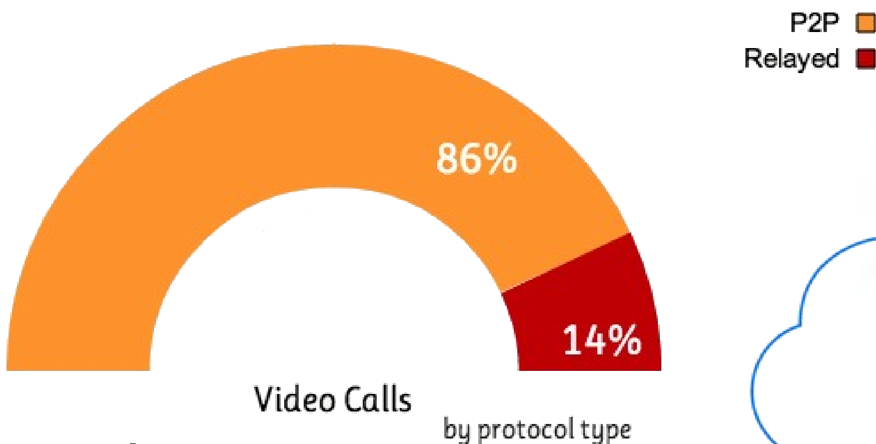
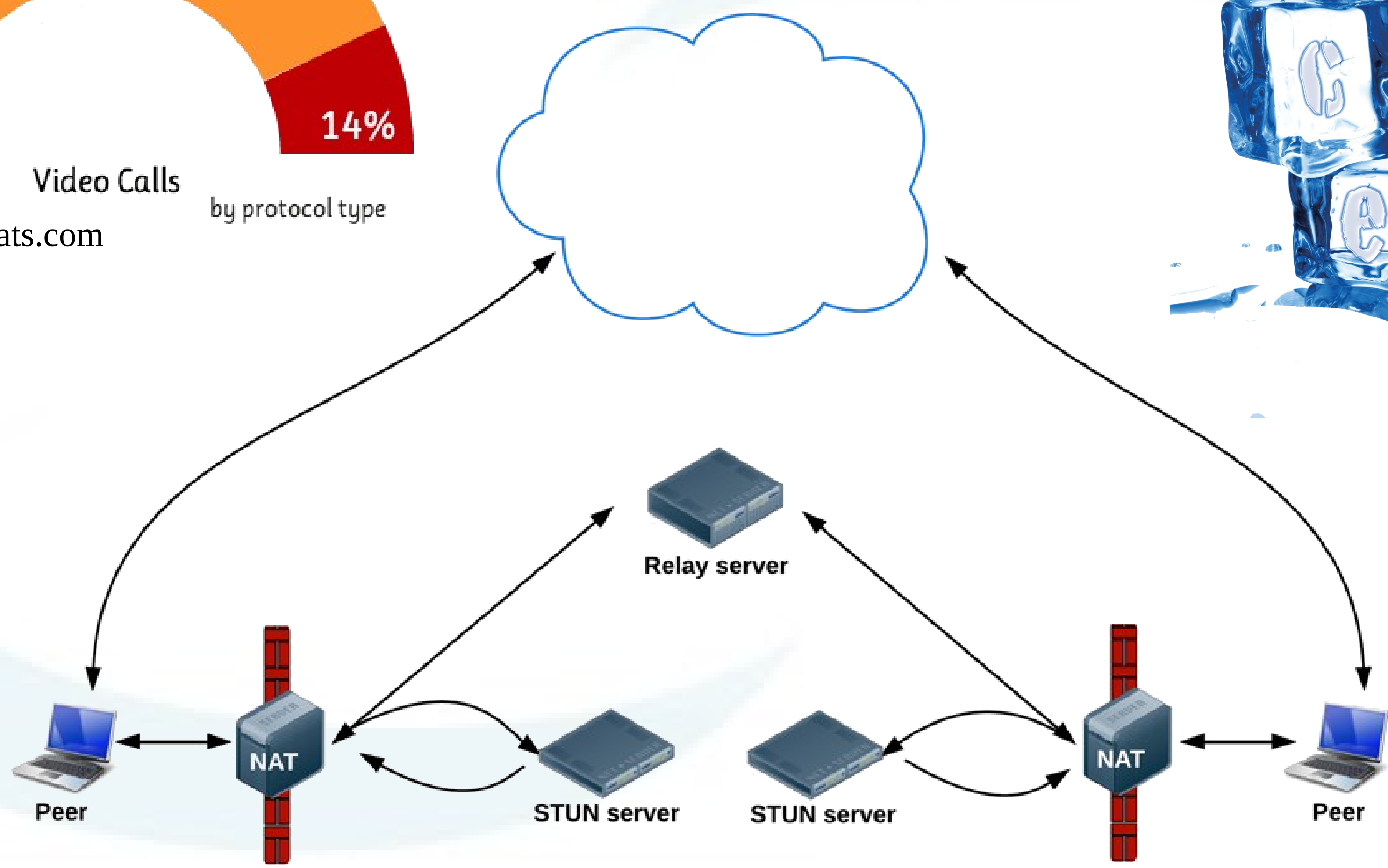


Image source: http://html5rocks.appspot.com/en/tutorials/webrtc/basics/jsep.png

Interactive Connectivity Establishment (ICE)



Source: webrtcstats.com

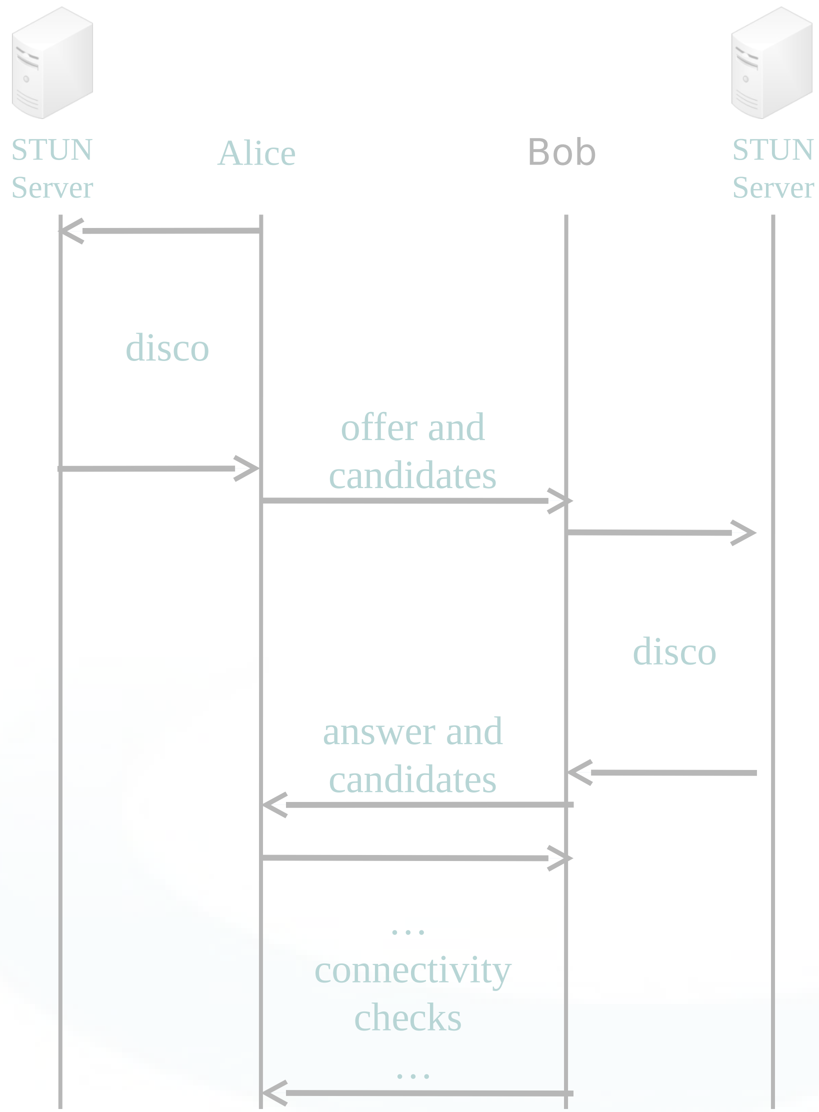


Standard Based Firewall/NAT Traversal

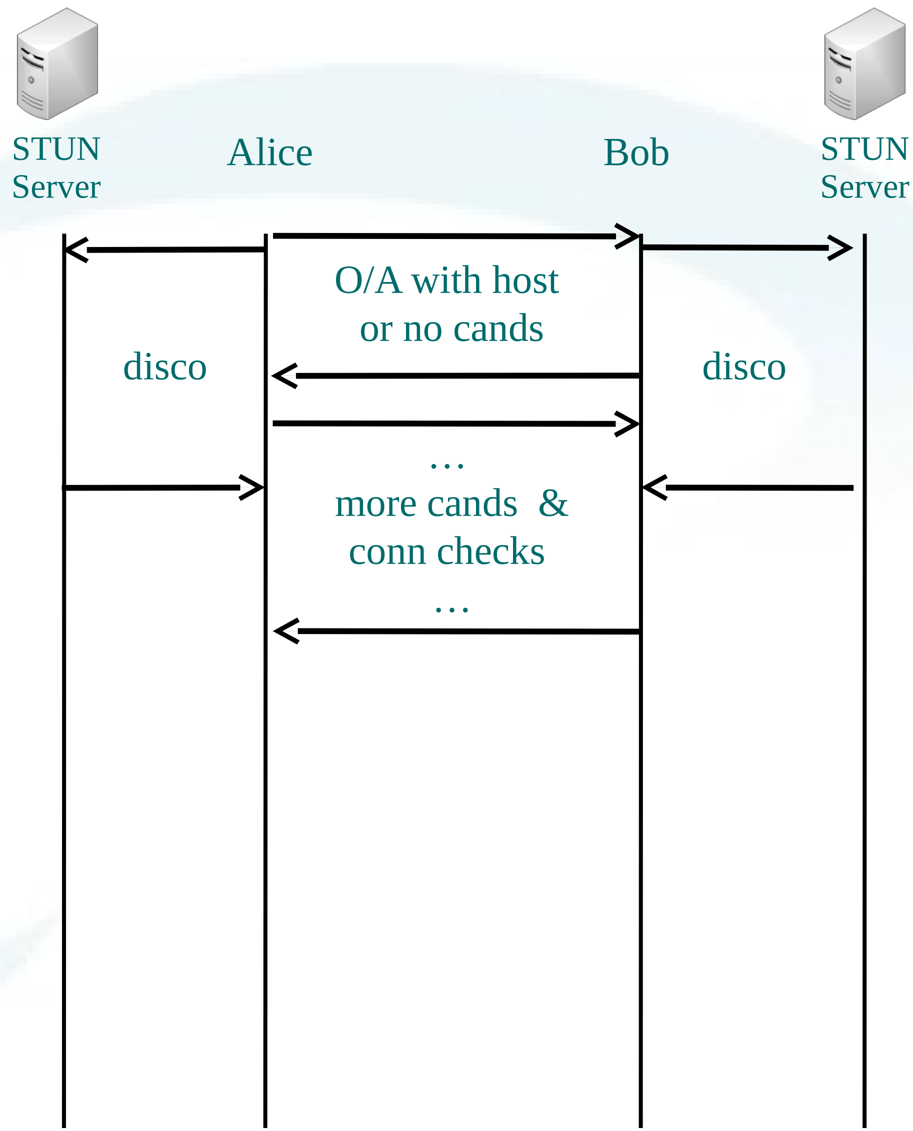
- ICE RFC5245 (STUN/TURN)
  - Tries to find the best path
  - Firewall traversal
  - IPv4, IPv6 Inter-working
  - Multiple IP addresses
- Beyond ICE
  - RFC5245 drawback
    - lengthy
  - Trickle ICE draft
    - Reducing session establishment time
    - Reducing ICE processing times
  - Incremental Provisioning of Candidates for the Interactive Connectivity Establishment (ICE) Protocol
  - XMPP XEP-0176
    - Implemented

ICE vs. Trickle ICE

Slide from: trickle-ice-iet86-orlando.pptx



Vanilla ICE as per RFC 5245



Technically

- WEBRTC JavaScript API
- WebRTC does not specify a signaling protocol
  - Designed in mind SIP, XMPP/JINGLE compatibility,
  - WebRTC signaling is fully application specific
- WebRTC is about Media handling
  - DTLS-SRTP (Audio, Video)
  - SCTP over DTLS (Data)
- NAT / Firewall traversal
  - IPv4/IPv6
  - Multiplexing data/media
- Security
  - Identity,Encryption, Privacy
- Congestion Control

- IETF RTCWEB WG
- Fresh / Current / Bleeding edge IETF standards
  - SDP media bundling
  - ICE (STUN/TURN)
    - Trickle ICE
  - DTLS-SRTP (~~SDES-SRTP~~)
  - RTP SAVPF
    - Secure RTP
    - RTCP feedback
  - RTP RTCP multiplexing
  - RTP multiplexing (audio video)
  - codecs (e.g. VP8, Opus, etc.)
  - SCTP (data channel)
  - DTLS-SRTP (video, audio)
- Implementing fresh new standards cause compatibility issues

Architecture Overview

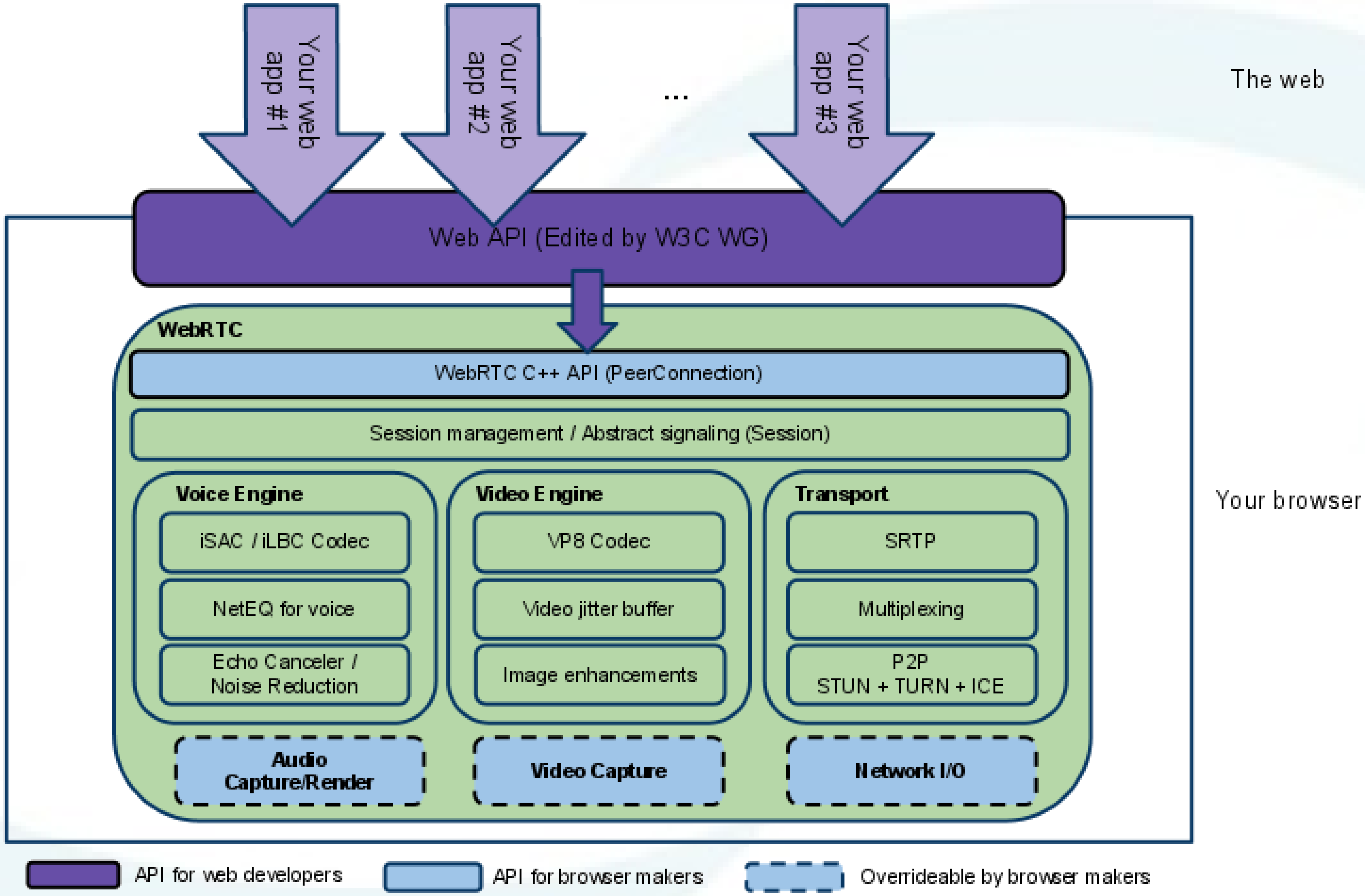


Image source: http://www.webrtc.org/_/rsrc/1317202919504/reference/WebRTCpublicdiagramforwebsite%20%282%29.png

Protocol Stack

- Peer-to-Peer media communication
- RTCP Multiplex
- Media Multiplex (audio, video)

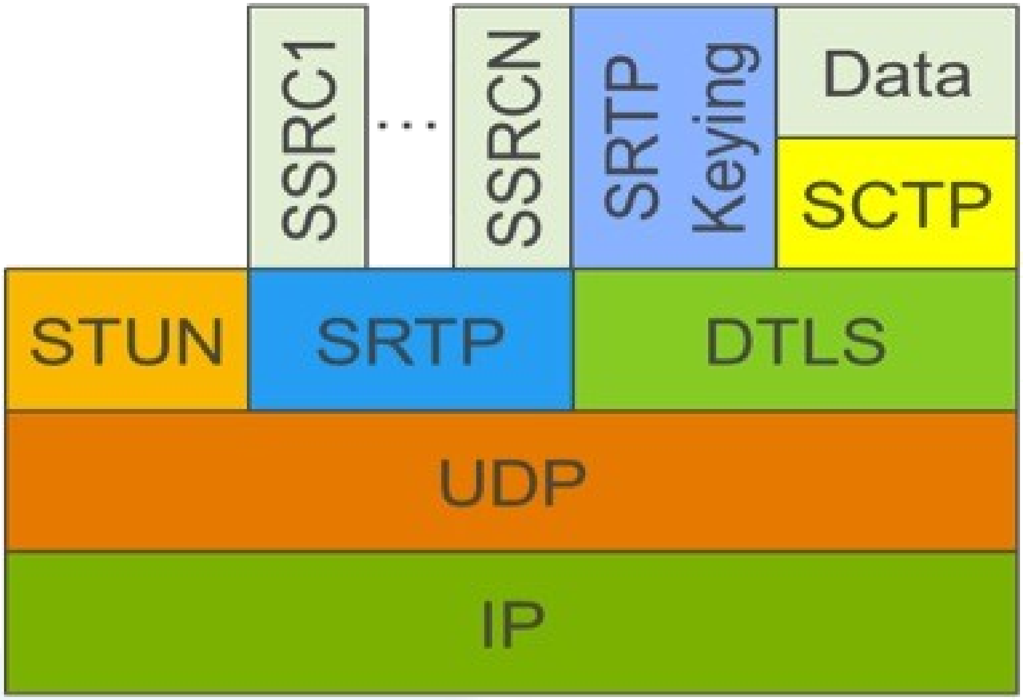


Image source: http://www.sloreto.com/slides/Aalto022013WebRTC/images/protocolStack.jpg

Security

- Trust in your browser only
- Secure End to End Communication
- GetUserMedia
  - Secure User Interface opt-in (e.g. Camera, audio access)
  - User can allow/deny audio video source usage
- Media/Data Encryption is mandatory!
  - DTLS-SRTP / DTLS
  - SDES-SRTP
    - “MUST NOT implement” according IETF 87
- AAI identity provision
  - WebRTC Security framework
  - SDP attached Identity Assertion (a=identity: base64)
- Signaling is not defined by WebRTC
  - e.g. SIP over WSS(TLS+Websocket)

Codecs

- Audio
  - Opus (royalty free, RFC 6176)
    - supports constant and variable bitrate encoding from 6 kbit/s to 510 kbit/s, frame sizes from 2.5 ms to 60 ms, and various sampling rates from 8 kHz (with 4 kHz bandwidth) to 48 kHz (with 20 kHz bandwidth, where the entire hearing range of the human auditory system can be reproduced)
  - iSAC (internet Speech Audio Codec)
    - a robust, bandwidth adaptive, wideband and super-wideband voice codec developed by Global IP Solutions
  - iLIBC (internet Low Bitrate Codec RFC 3951) narrowband voice codec
    - free narrowband voice codec that was developed by Global IP Solutions
  - G.711 (alaw/ulaw)
- Automatic Gain Control (AGC)
- Acoustic Echo Cancellation (AEC)
- Video
  - VP8 Chrome, Firefox
  - H.264 Browser(Ericsson Lab), (Firefox planed)
- VoiceEngine, VideoEngine, NetEQ, AEC, etc all stem from the GIPS acquisition

Battle for Mandatory To Implement(MTI) Video Codec

- Battle for WebRTC mandatory to implement (MTI) codec
- Audio MTI codecs
  - G.711 (alaw/ulaw)
  - Opus
- Video (?!)
- Google
  - Hangout H.264=>VP8
  - Chrome only VP8/VP9 support
- Cisco (just announced)
  - Cisco will open H.264 codec
  - Cisco will pay MPEG LA
  - Mozilla will support Cisco binary H.264 codec
  - http://www.openh264.org/

- video codec proposals, and backers
  - VP8 (VP9)
    - Google
  - H.264 (H.265)
    - Ericsson
    - Nokia
    - BlackBerry
    - Qualcomm
    - Orange
    - Cisco
    - Microsoft
    - Apple
- Both has Pros & Cons

WEBRTC API

- Major API Components
  - GetUserMedia
    - Acquiring audio and video
    - which allows a web browser to access the camera and microphone
  - PeerConnection
    - P2P Communication
      - Codec negotiation, Security
      - Media handling, Bandwidth Management
      - etc.
  - DataChannels
    - which allow browsers to share data via peer-to-peer
  - Peer-to-peer DTMF
  - RTCStatsReport
  - Identity

Architecture Overview

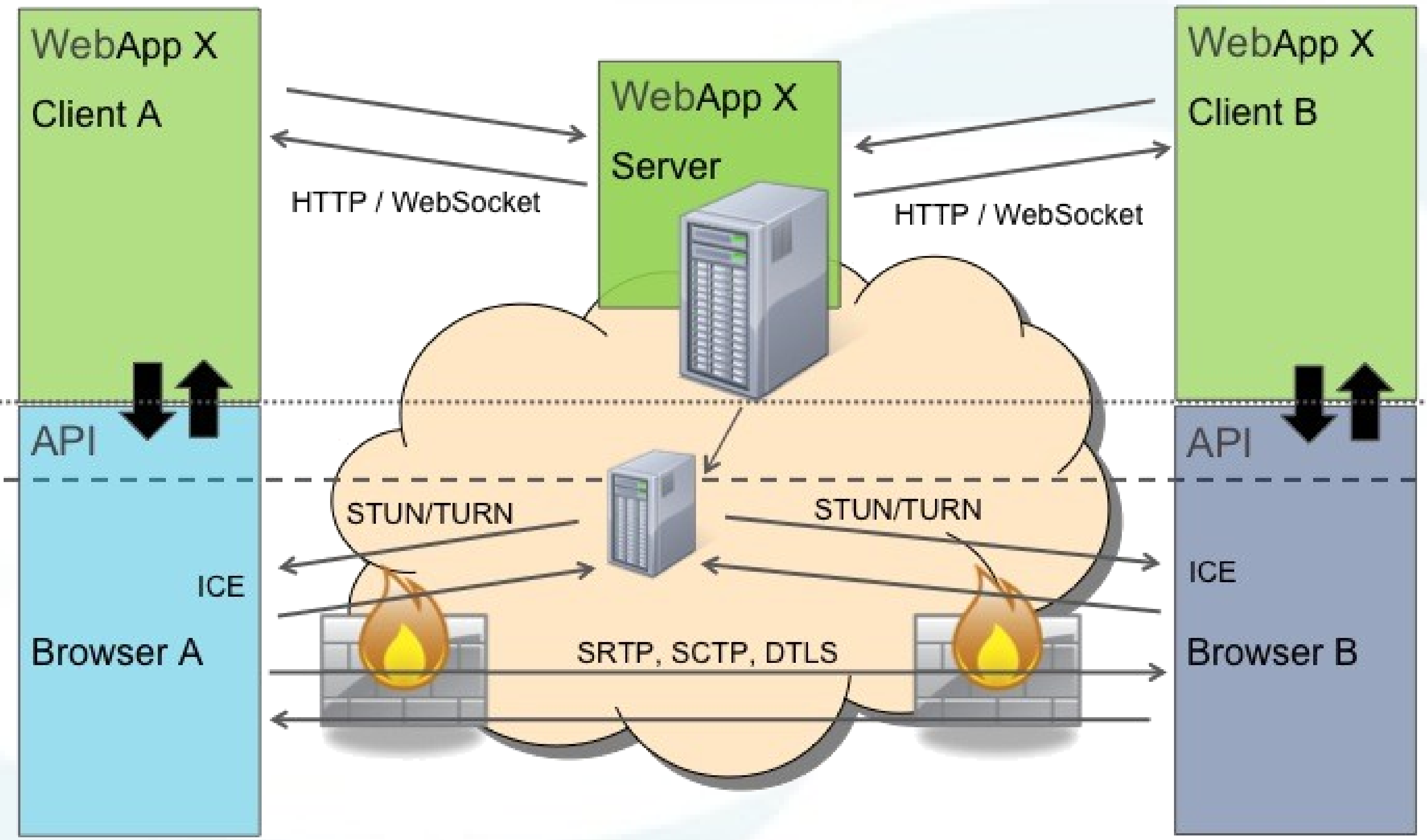
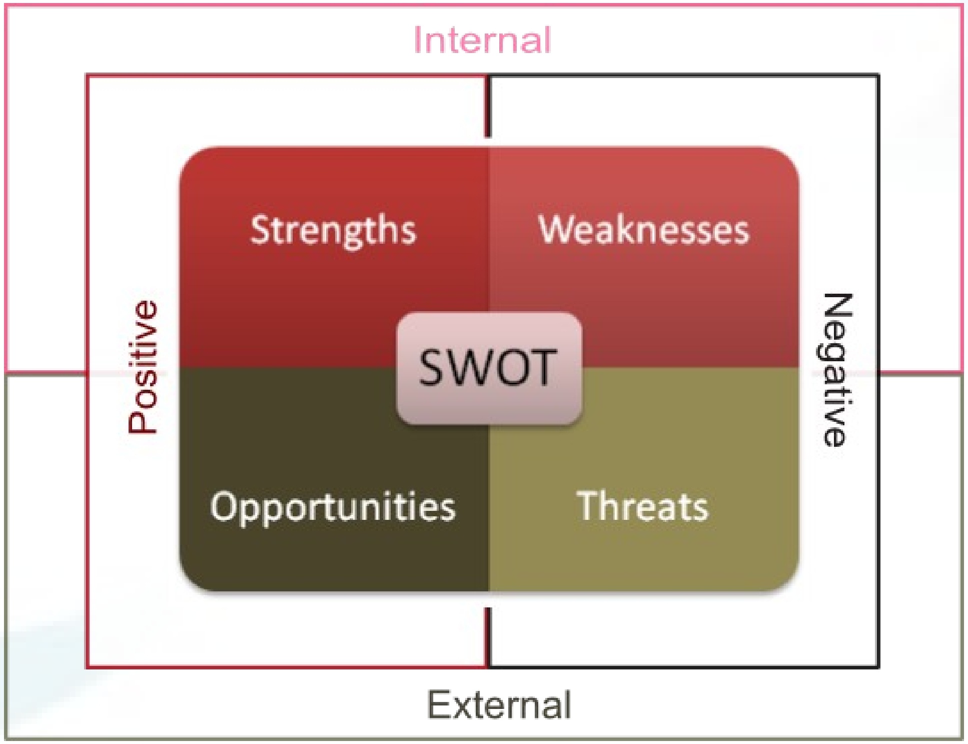


Image source: http://www.sloreto.com/slides/Aalto022013WebRTC/images/WebRTC_Architecture0.jpg

WebRTC SWOT analysis

- SWOT analysis:
  - Strengths
  - Weaknesses
  - Opportunities
  - Threats



SWOT: Strengths

- No plugins
  - No Flash, Java, Silverlight etc. needed
- Client deployed everywhere
  - No sw client install needed:
    - 1000000000+ WebRTC endpoints
  - Client is always up2date. (Browser auto updates)
- Multi Platform
  - PC
  - Phone, Tablet
  - And more e.g. any HW Android could run
- Security is mandatory
  - peer-to-peer
- HD video,
- Wideband audio
- Opt-in Privacy
- Open
  - Source
  - Standards based
  - Nothing proprietary(?)
- Web Multimedia
  - Voice
  - Video (webcam, screencapture)
  - Data
- Standard based Firewall/NAT traversal
  - ICE (STUN/TURN)
  - IPv6 and IPv4 negotiation, interoperability
- WebRTC is part of HTML5
  - Web JS API is simple and hides complexity
- Implementations
  - Chrome,Firefox

SWOT: Weaknesses

- Early adopters phase
  - not mature final standard (draft), rapidly changing implementations
- Browser implementation compatibility
- Depends other sw infrastructure operations
  - STUN/TURN server, MCU, Gateway
- Web Application is not native (limitations)
  - e.g. Facebook and HTML5
- Codec War (H.264 vs. VP8) future (H.265 vs. VP9)
- Many different emerging approaches for Signaling
  - JSON over WebSocket (Fall-back to BOSH, COMET)
  - SIP over WebSocket
  - etc…
- WebSocket is not yet implemented in every HTTP proxy.
- Desktop sharing, statistics is not yet implemented in every browser

SWOT: Opportunities

- WebRTC Hype
- HTML5 (WebRTC) as an universal application platform.
  - Disturbing communication market
- Transparent Standard based secure platform for RTC
- New possibilities / New applications
  - Games, Video support call center, Lecture Recording
  - Apps Mobile, Tablet /Android/
  - Etc.
- Videoconference to anyone who has a browser
  - Billion installed/updated clients.
- Bridge between Telco and Web world
- Trusted, Open Source peer to peer communication
  - AAI integration
- Next gen video codecs: e.g. VP9 (SVC)

SWOT: Threats

- Compatibility with exiting RTC vendors implementations
  - WEBRTC implements leading edge IETF standards
    - current installed videoconference / telepresence room don't.
  - Codec MTI war!
- Browser implementation, compatibility
  - Internet Explorer, Opera etc..
  - Adapter.js
- Not defined Endpoint/User Identification
  - URI, E.164, etc.
  - Lack Unified Directory / Phone-book
- Mobile adaptation (iOS)
- Legal Issue: Communication Regulation
  - Lawful interception
- Too many signaling protocols can lead to Walled Gardens compatibility issues.
- Alternative APIs (ORTC, CU-RTC-WEB)

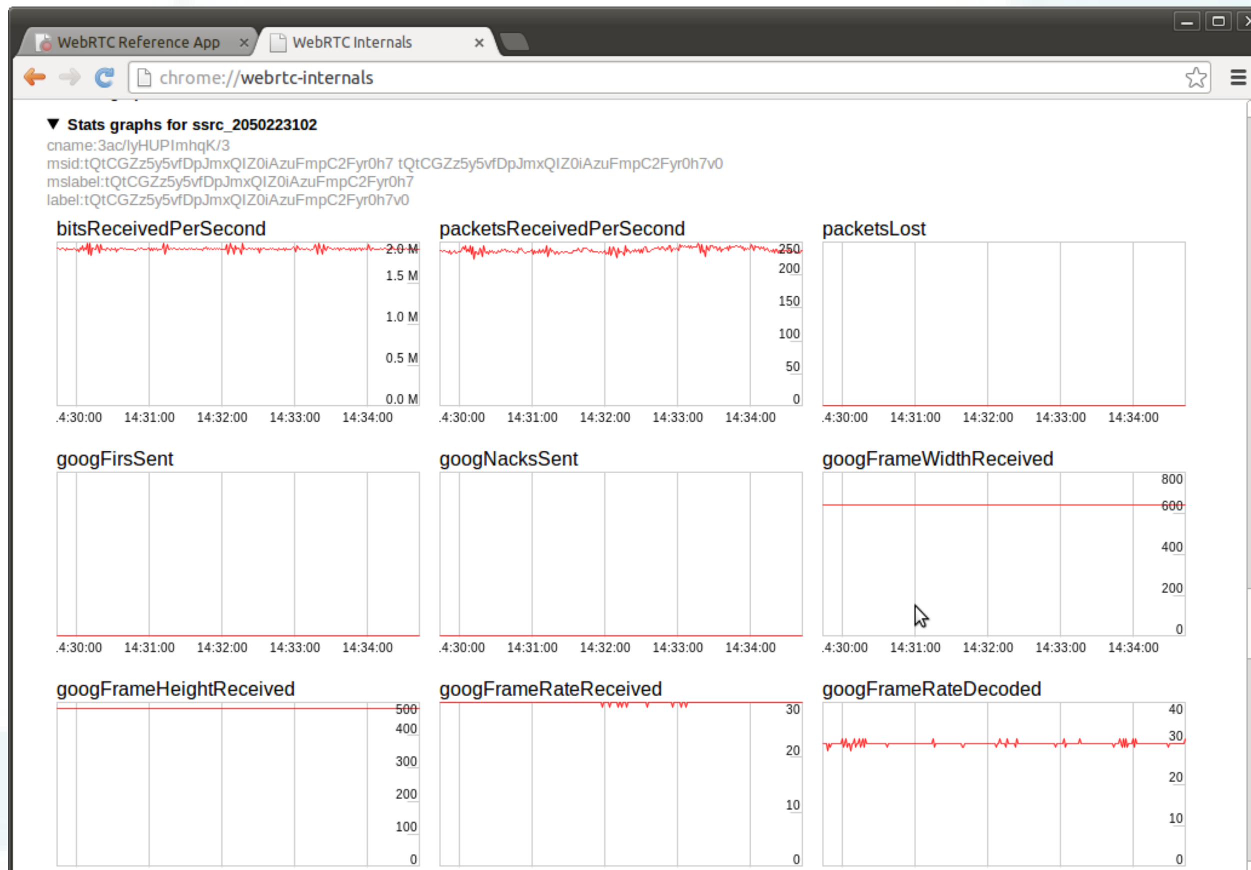
WebRTC and NREN's

- TNC2013 TERENA Technical Advisory Council
  - Jan Meier: WebRTC Why you should care?
- Big Blue Button WebRTC Support
  - Funded by UNINET, NorduNet
- 2013 Aug 26 WebRTC meeting
  - Big Blue Button WebRTC support (NORDUNET)
  - Videoconference Gateway/MCU (NIIFI, JANET)
  - Lecture Recording (REDIRIS)
- Tests
  - WebRTC Gateway (SIP,XMPP,H323 etc.)
  - WebRTC MCU
- Open Mailinglist
  - discussion@nrenum.net
  - webconf@terena.org

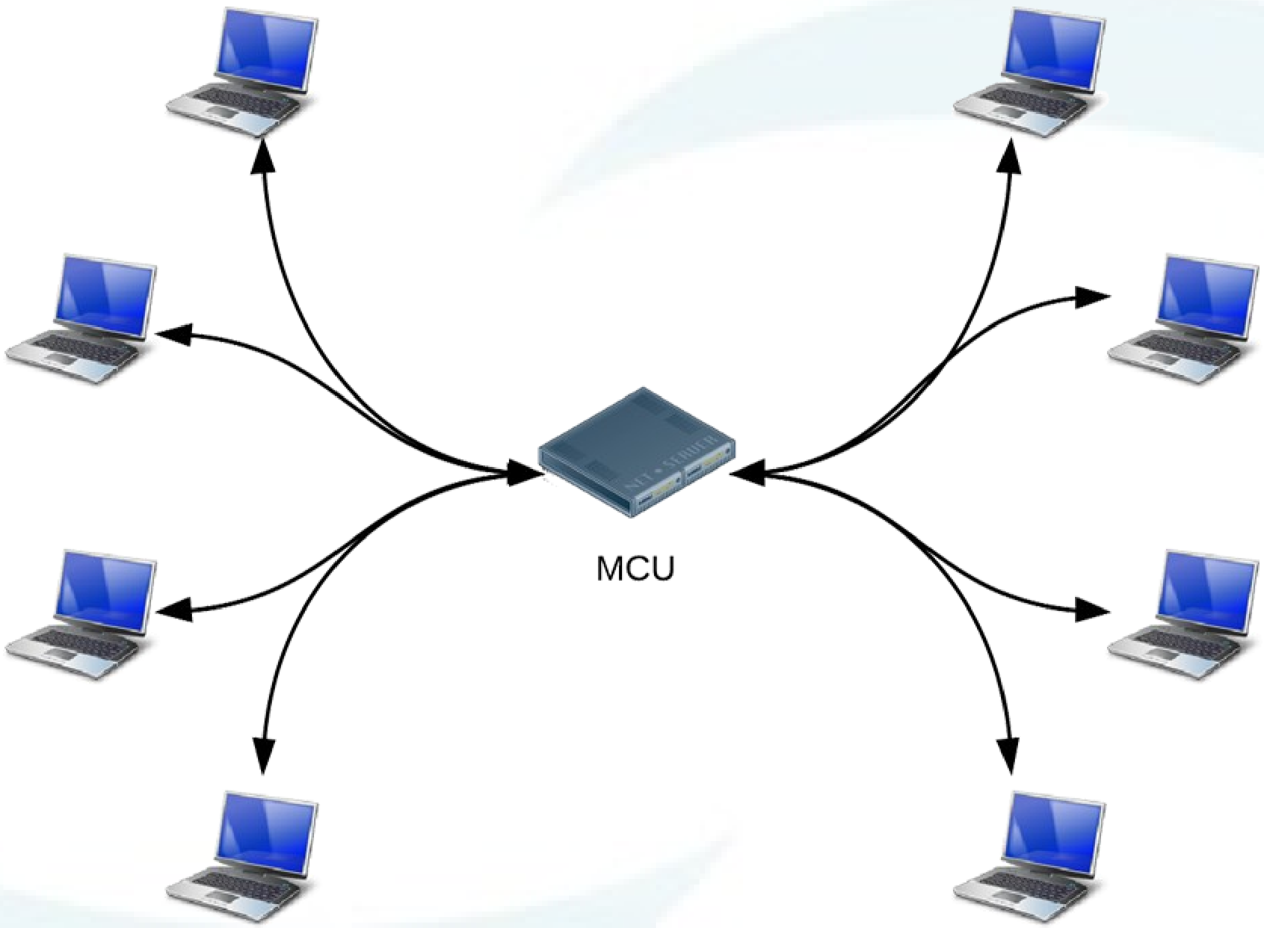
TERENA

Diagnostic / Interop

- Browser Interop http://www.webrtc.org/interop
  - https://code.google.com/p/webrtc/source/browse/trunk/samples/js/base/adapter.js
- Developer / Diagnostic tool
  - chrome://webrtc-internals
  - Firefox planed



MCU



MCU, Gateway

- MCU
  - WebRTC is about Peer2Peer
  - So limited Multipoint capabilities
  - WebRTC endpoint need an MCU for large N-way calls

- Gateway/SBC
  - Interoperability
    - RTP
      - SDES-SRTP
      - DTLS-SRTP
      - RTP
    - Demultiplex
      - RTCP
      - Media channel
    - SAVPF<=>AVP
      - RTCP feedback
    - ICE(STUN/TURN)
  - Security
  - Transcoding Video, Audio
    - e.g. VP8 <=> H.264

Webrtc MCU vendors

- Open Source
  - http://www.medooze.com/products/mcu/functionality.aspx
    - Argentinian universities VoIP workgroup has been using for about a year.
    - http://www.youtube.com/watch?v=pocgfJXmwV4 (in Spanish)
  - http://lynckia.com/
  - http://code.google.com/p/telepresence/
    - NIIFI tested
- Commercial
  - http://www.requestec.com/site/platform/architecture.jsp
  - http://acano.com/tour/
  - http://www.pexip.com/requirements
    - NIIFI tested
    - Version 2
    - SRTP-DTLS (coming V3)

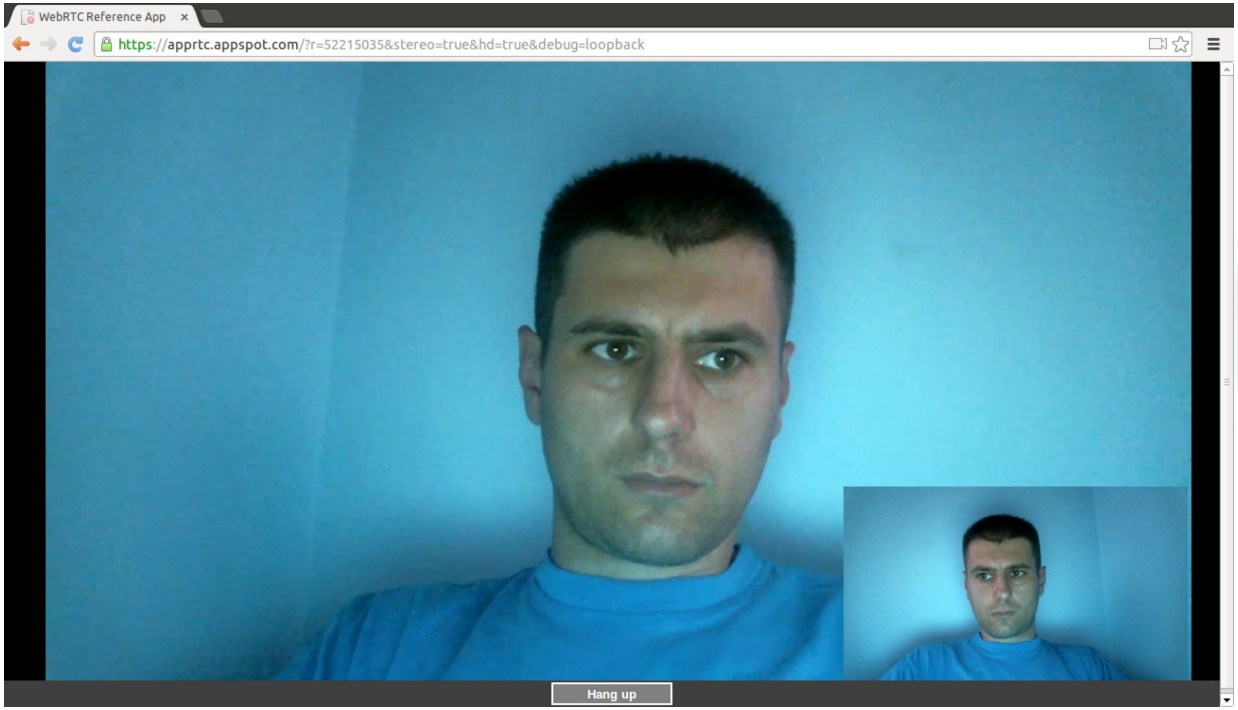
Open Source implementations

- IP PBX
  - FreeSwitch
    - SIP over Websocket
    - SRTP-DTLS (git version)
    - video transcoding fs-video branch
  - Asterisk
    - SIP over Websocket
- SIP Proxy
  - Kamailio
    - SIP over WebSocket
  - OverSIP
    - SIP over WebSocket
- RTP PROXY
  - mediaproxy-ng
- JS client library
  - Doubango SIPML5
  - JsSIP
- Gateway
  - Doubango webrtc2sip (GW)
- Web Conferencing
  - Big Blue Button

WebRTC API vs Alternative APIs

- Current nearly 1.0 WebRTC API couldn't be perfect.
  - World Wide consensus is big challenge.
  - First make API stable.
  - Redesign takes time. So redesign only after stable API 1.0
  - http://dev.w3.org/2011/webrtc/editor/webrtc.html
  - http://dev.w3.org/2011/webrtc/editor/getusermedia.html
- API Alternatives
  - WebRTC Object API (ORTC)
    - https://rawgithub.com/openpeer/ortc/master/ortc.html
    - http://www.w3.org/community/orca/
  - Microsoft (CU-RTC-Web)
    - http://lists.w3.org/Archives/Public/public-webrtc/2012Aug/0014.html

Demo



- https://apprtc.appspot.com/
- Options:
  - stereo=true
  - hd=true
  - debug=loopback
  - video=
  - audio=
  - ss= (stun)
  - st=(turn)
  - For more parameters see:
    - https://code.google.com/p/webrtc/source/browse/trunk/samples/js/apprtc/apprtc.py)
  - Loopback test

    https://apprtc.appspot.com/?r=52215035&hd=true&debug=loopback
- http://tryit.jssip.net/
  - Sip account needed (e.g. http://vvc.niif.hu)

chrome media constraints

- Audio
  - Google Specific
    - googEchoCancellation,
    - googEchoCancellation2,
    - googAutoGainControl,
    - googAutoGainControl2,
    - googNoiseSuppression,
    - googHighpassFilter,
    - googTypingNoiseDetection,
    - googAudioMirroring
- https://code.google.com/p/webrtc/source/browse/trunk/talk/app/webrtc/localaudiosource.cc
- Video Constraints
  - Common Options
    - MinAspectRatio, maxAspectRatio
    - MaxWidth, minWidth
    - MaxHeight, minHeight
    - MaxFrameRate, minFrameRate
  - Google Specific
    - googNoiseReduction,
    - googNoiseReduction,
    - googCpuOveruseDetection
  - https://code.google.com/p/webrtc/source/browse/trunk/talk/app/webrtc/videosource.cc

Links

- WebRTC
  - http://www.youtube.com/watch?v=p2HzZkd2A40
  - http://www.youtube.com/watch?v=E8C8ouiXHHk
  - http://vimeo.com/47682405
  - http://www.html5rocks.com/en/tutorials/getusermedia/intro/
  - http://www.html5rocks.com/en/tutorials/webrtc/basics/
  - http://io13webrtc.appspot.com
  - http://webrtchacks.com/signalling-options-for-webrtc-applications/
  - http://webrtchacks.com/webrtc-video-codec-discussion/
  - http://webrtchacks.com/webrtc-must-implement-dtls-srtp-but-must-not-implement-sdes/
  - http://prezi.com/qwejmltpng8x/webrtc/
  - http://www.slideshare.net/janmeijerno/tnc2013-tacmeijerwebrtcclean
  - http://www.youtube.com/watch?v=0gG2Y655Kk8
  - http://www.youtube.com/watch?v=jKz-Fjbw2mA
  - http://www.youtube.com/watch?v=Hzf6wsfnMKs
  - http://www.sloreto.com/slides/Aalto022013WebRTC/slides.html
- ICE / Trickle ICE
  - http://tools.ietf.org/html/draft-rescorla-mmusic-ice-trickle-00
  - http://www.ietf.org/proceedings/87/slides/slides-87-mmusic-11.pdf
  - https://github.com/emcho/trickle-ice/tree/master/slides
  - http://videotorium.hu/hu/recordings/details/4110,ICE_TURN_STUN_a_szabvanyos_media_tuzfalatjarasi_technologia

Links++

- Tutorial / Demo
  - https://www.webrtc-experiment.com/
  - http://idevelop.ro/ascii-camera/
  - http://webcamtoy.com/app
  - https://code.google.com/p/webrtc/source/browse/#svn%2Ftrunk%2Fsamples%2Fjs%2Fdemos%
  - https://vline.com/
  - http://webrtc.bigbluebutton.org/
  - https://opentokrtc.com/
  - http://mozilla.github.io/webrtc-landing/
  - https://webrtc.freeswitch.org/sipml5/
- Javascript SIP
  - http://www.jssip.net/
  - https://code.google.com/p/sip-js/
  - http://qoffeesip.quobis.com/
  - http://sipml5.org/

Interesting WebRTC related drafts

- RTC web Workgroup Page
  - http://www.webrtc.org/
  - http://tools.ietf.org/wg/rtcweb/
  - http://dev.w3.org/2011/webrtc/editor/webrtc.html
  - http://dev.w3.org/2011/webrtc/editor/getusermedia.html
- Drafts
  - http://tools.ietf.org/html/draft-ietf-rtcweb-overview-08
  - http://tools.ietf.org/html/draft-ietf-rtcweb-rtp-usage-10
  - http://tools.ietf.org/html/draft-ietf-avtcore-multi-media-rtp-session-03
  - http://tools.ietf.org/html/draft-ietf-rtcweb-security-05
  - http://tools.ietf.org/html/draft-ietf-sipcore-sip-websocket-09
  - http://tools.ietf.org/html/draft-ietf-rtcweb-security-arch-07
  - http://tools.ietf.org/html/draft-jennings-rtcweb-signaling-01

Thank You!