

# *Deploying WebRTC:* *“Would you like it well-tested?..”*

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# Major Areas of RTC Testing

	Signaling	Signaling + Media
<b>Functional Testing / Interoperability Testing</b>	<ul style="list-style-type: none"><li>• Singular calls are sufficient (typically)</li><li>• Manual testing / Test automation with scripts</li></ul>	
<b>Robustness Testing / Security Testing</b>	<ul style="list-style-type: none"><li>• Singular calls are sufficient (typically)</li><li>• Manual testing / Test automation with scripts</li><li>• Randomized inputs (fuzz testing)</li></ul>	
<b>Load Testing / Stress Testing</b>	<ul style="list-style-type: none"><li>• Many concurrent calls</li><li>• Usually driven by scripts or predefined flows</li><li>• Monitoring and measuring</li></ul>	

# Some WebRTC Testing Specifics

- Cross-browser interop. (*specs compliance, codecs*)
  - *Esp. before the official standard, MTI video codec(s), etc.*
- Testing signaling-server function
  - *Depends on specific signaling protocol(s) being used*
- Testing media-server function, *if there is one...*
  - *Do you do media transcoding? Conferencing?*
- TURN-server load testing, *if relying on TURN...*

# Your mileage may vary...

- Using WebRTC-based 3<sup>rd</sup>-party services/frameworks
  - *Does your supplier offer help with testing or test tools?*
  - *Can you “skip” some test work thanks to SLA guarantees?*
- Testing with separate commercial products/services
- Opting for “DIY-testing”

# (Web)RTC Test Automation Parts

## Test-scenario logic (*e.g. with scripts*)

### Call / Session signaling

- SIP, XMPP *and alike*
- JSON-based
- REST-based
- “Project WONDER”

### Media signaling

- SDP (WebRTC 1.0)
- *ORTC-based*  
(*post-WebRTC 1.0*)

### Media payload(s)

- Audio / Video
- Data, in needed format(s)

### Auxiliary functions

- Logging
- Test Helpers
- *etc.*

## Basic RTC Mechanisms / Platform & Network Interactions

# Browser-based Test Automation

- Use fake media devices in `getUserMedia()`
  - Start Chrome with **`--use-fake-device-for-media-stream`**
    - You may also add **`--use-file-for-fake-video-capture=sample.y4m`**
  - In Firefox: **`getUserMedia( { video: true, audio: true, fake: true }, ...)`**
- Disable permission dialogs for camera/mic access
  - Start Chrome with **`--use-fake-ui-for-media-stream`**
  - In Firefox preferences: **`media.navigator.permission.disabled:true`**  
(Apparently, FF 33 auto-disables the dialogs with “fake: true” ???)

# Browser-based Test Automation

- Scripting and testing UI interactions
  - *Selenium (<http://www.seleniumhq.org/>)*
- Running browsers in headless mode
  - *Linux and OS X: X virtual framebuffer (Xvfb / Xdummy)*
  - *As an example, check scripts from Otalk:  
<https://github.com/otalk/webrtc-tester>*

# Node.js-based Test Automation

- node-webrtc
  - *<http://js-platform.github.io/node-webrtc>*
  - *WebRTC peer connections and data channels in Node.js!*
  - *Media stream objects are just “stubs”, as of now...*
- “Missing pieces” (*work in progress, stay tuned! ☺*)
  - *Better tools for scripting signaling scenario tests*
  - *Media stream emulation (predefined or file-based)*

# Signaling Traces with WebRTC

- Wireshark *is* your good friend!
  - *Limitations: Encrypted data, using pcap in 3rd-party apps*
- Collecting traces in signaling libs: *to be improved*
- Traces in browsers: *evolving, but still in early days*
- ***Proposing a common format for signaling archives***
  - ***“Simple Application-Level-Signaling Archive” (SALSA)***
  - ***<https://github.com/VladimirTechMan/salsa-format-spec/>***

# Thank you for your attention!

Questions and feedback are welcome!



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