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ClueCon

Developing RTC Applications with Microsoft Edge

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What Can You Do With Edge?

Build ORTC applications

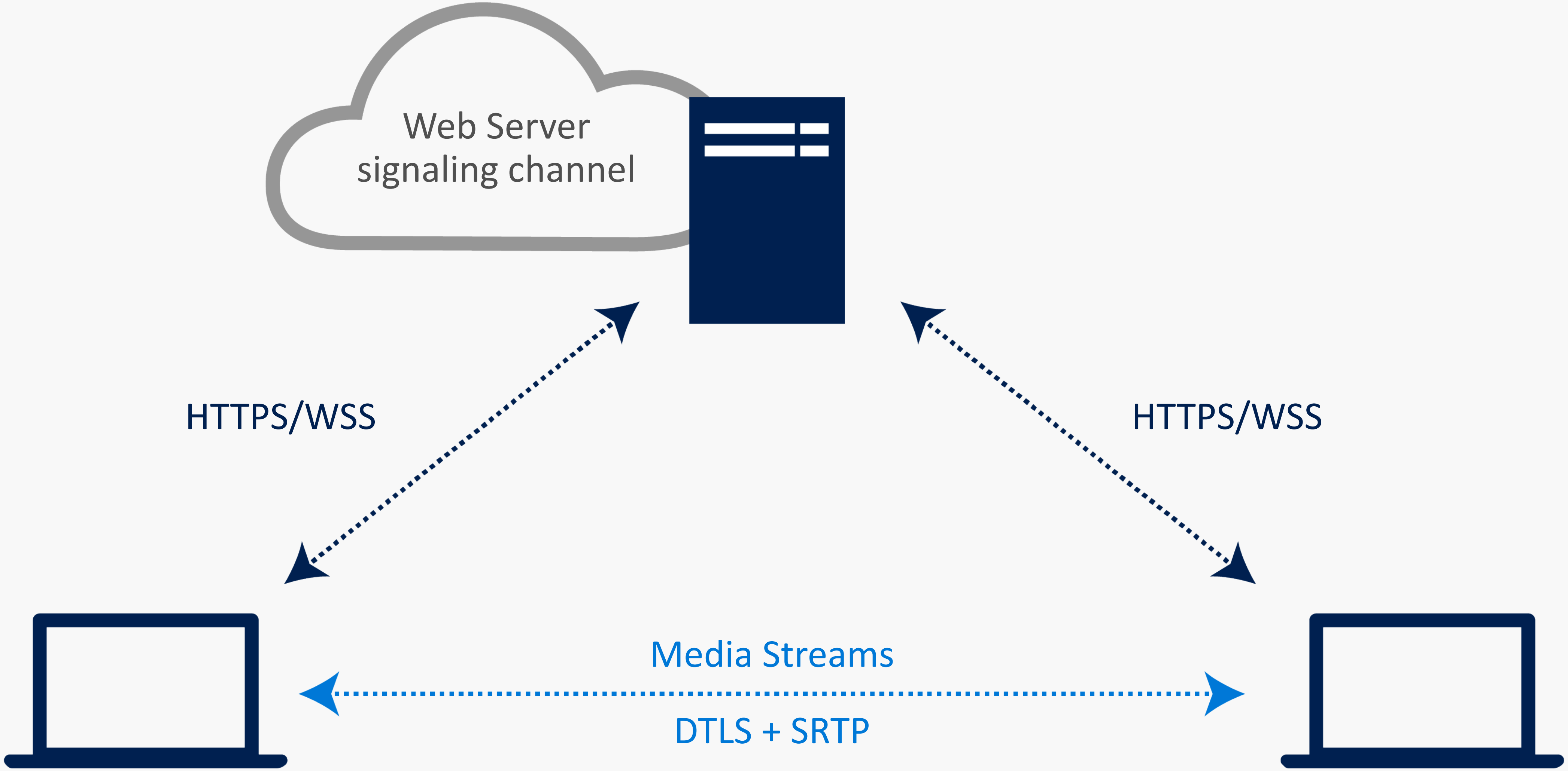
Run WebRTC 1.0 Applications

- Using the current WebRTC 1.0 API over ORTC (adapter.js)
- Using legacy native WebRTC 1.0 API

Step-by-Step: Building an ORTC Application

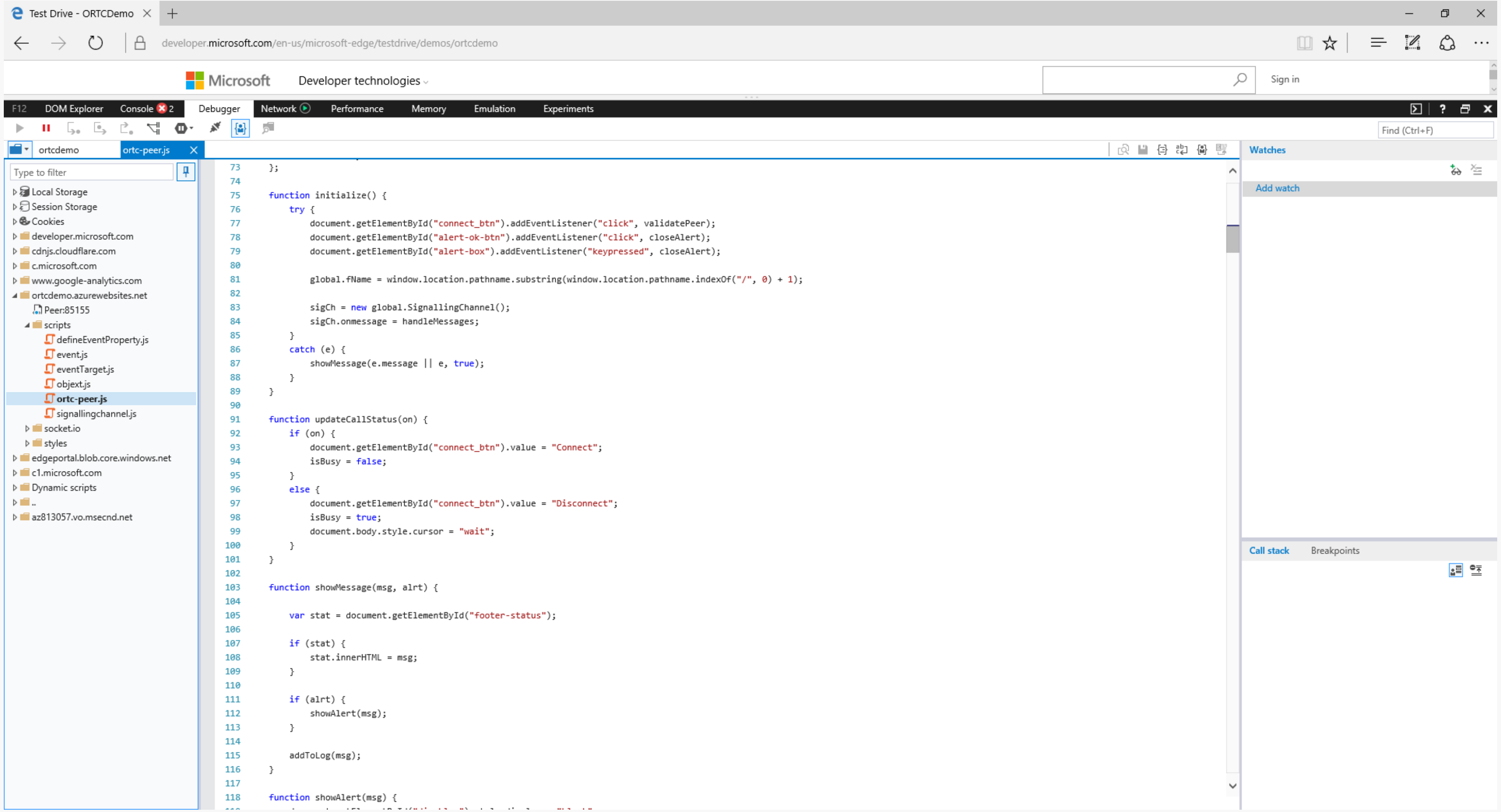
1. Establish a signaling channel
2. Create MediaStream object with audio and video tracks
3. Create transports and exchange parameters.
4. Create sender and receiver objects and extract and exchange capabilities
5. Start the ICE and DTLS transports, call send() and receive() methods
6. Connect incoming media tracks from RtpReceiver to audio and video tags

A Simple Topology



ORTC Demo Application

https://developer.microsoft.com/en-us/microsoft-edge/testdrive/demos/ortcdemo/



App level code flow

// 2. Create MediaStream object (i.e. Media Capture API) with one audio track and one video track

```
navigator.mediaDevices.getUserMedia({
      "audio": true,
      "video": {
         width: 640,
         height: 480,
         facingMode: "user"
      }
   }).then(
      gotMedia
    ).catch(
      gotMediaError
    );
```

// 3. Create transports (ICE gatherer, ICE transport, DTLS transports, etc.) and exchange parameters.

```
function initiateConnection() {
   updateCallStatus();

   var iceOptions = { "gatherPolicy": "all", "iceServers": [{ "urls": "turn:turn-
      testdrive.cloudapp.net:3478?transport=udp", "username": "redmond",
      "credential": "redmond123" }] };

    iceGathr = new RTCIceGatherer(iceOptions);
    iceTr = new RTCIceTransport();
    dtlsTr = new RTCDtlsTransport(iceTr);
...
   signalMessage(JSON.stringify({
      params: {
                "ice": iceGathr.getLocalParameters(),
                "dtls": dtlsTr.getLocalParameters()
              }
   }));

...
```

// 4. Create sender and receiver objects and extract and exchange capabilities

```
function gotMedia(stream) {
   var audioTracks = stream.getAudioTracks();

   if (audioTracks.length > 0) {
      var audioTrack = audioTracks[0];
      local_audio_MST = audioTrack;

      audioSender = new RTCRtpSender(audioTrack, dtlsTr);
      sendAudioCaps = RTCRtpSender.getCapabilities("audio");

      signalMessage(JSON.stringify({
         params: {
            "sendAudioCaps": sendAudioCaps
         }
      }));
}
...
```

// 5. Start the ICE and DTLS transports

```
if (message.params) {
    var remote = message.params;

    if (remote.ice) {
        remoteIceParams = remote.ice;
        remoteDtlsParams = remote.dtls;

        if(localCandidatesCreated){
           iceTr.start(iceGathr, remoteIceParams,(selfInfo.dtlsRole && selfInfo.dtlsRol
           === "client" ? "controlled" : "controlling" ));
                 dtlsTr.start(remoteDtlsParams);
           }
    }

...
```

// 5. Call the send() and receive() methods
// 6. Connect incoming media tracks from RtpReceiver to audio and video tags

```
if(remote_audioRecvParams){
   var remote = remote_audioRecvParams;
   var audioRecvParams = util.myCapsToRecvParams(receiveAudioCaps, remote.sendAudioCaps);
   audioRecvParams.encodings.push(util.RTCRtpEncodingParameters(1001, 0, 0, 0, 1.0));
   audioReceiver.receive(audioRecvParams);
   trackCount++;
      if ( trackCount == 2) {
         videoRenderer.srcObject = renderStream;
      }
}
...
if( remote_audioSendParams ){
   var remote = remote_audioSendParams;
   var audioSendParams = util.myCapsToSendParams(sendAudioCaps, remote.receiveAudioCaps);
   audioSendParams.encodings.push(util.RTCRtpEncodingParameters(1001, 0, 0, 0, 1.0));
   audioSender.send(audioSendParams);
}
```

WebRTC 1.0 Support on ORTC

Using the adapter.js library, you can build audio and video applications that run on Edge, Chrome and Firefox using the latest WebRTC 1.0 API.

WebRTC 1.0 Application

adapter.js

ORTC

Latest release of adapter.js:
https://github.com/webrtc/adapter/blob/master/release/adapter.js
WebRTC 1.0 sample applications:
https://github.com/webrtc/samples

WebRTC 1.0 testbed:
https://github.com/fippo/testbed

What Applications Can Use adapter.js?

- adapter.js supports getUserMedia as well as P2P use of RTCPeerConnection, including:
  - Single-stream video applications supporting H.264/AVC and VP8
    - P2P audio or audio/video chat
  - Conferencing applications using an MCU
    - MCU receives multiple video streams from participants and outputs a composite video stream.
    - Avoids need for multi-stream support (not interoperable yet)
    - Example: FreeSwitch Verto Client
- Not supported yet
  - WebRTC 1.0 object model
  - Multi-stream audio or video (e.g. no Unified Plan)
  - Re-negotiation

RTCPeerConnection Demo (adapter.js)

- In November 2013, Sam Dutton of Google published a “WebRTC In the Real World” tutorial: https://www.html5rocks.com/en/tutorials/webrtc/infrastructure/
  - Tutorial example uses sockets.io and a node.js server to support a 1:1 audio/video session.
  - Due to changes in sockets.io and the https: requirement in Chrome getUserMedia, the original code no longer runs, but is available here:
  - http://simpl.info/rtcpeerconnection/
  - https://bitbucket.org/webrtc/codelab/src/master/complete/step5/
- Simon Pietro Romano updated the code and added data channel support, as available here:
  - https://github.com/spromano/WebRTC_Book
  - This code also no longer runs
- With a little “renovation” and addition of adapter.js, code now runs (and interoperates) on Chrome, Firefox and Edge.

server code (runs in node.js):

http://internaut.com:8080/~baboba/cluecon-tutorial/adapter/server.js

client code

http://internaut.com:8080/~baboba/cluecon-tutorial/adapter/js/main.js

client renovations

- Provided a TURN server URI in IceServers
  - Edge does not support STUN URIs.
- New promise-based APIs
  - navigator.mediaDevices.getUserMedia
  - createOffer/createAnswer
  - setLocalDescription/setRemoteDescription
  - addIceCandidate
- End-of-candidates handling
  - Edge: calling addRemoteCandidate on a null event.candidate required for ICE processing to start.
  - adapter.js now emits a special “end-of-candidates” candidate
- Not fixed yet: removal of deprecated APIs (generates warning in Firefox)
  - addStream (in favor of addTrack)
  - onaddstream (in favor of ontrack)

Questions?