

# Web 2.0

## Lecture 7: Protocols for the Realtime Web

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## Overview

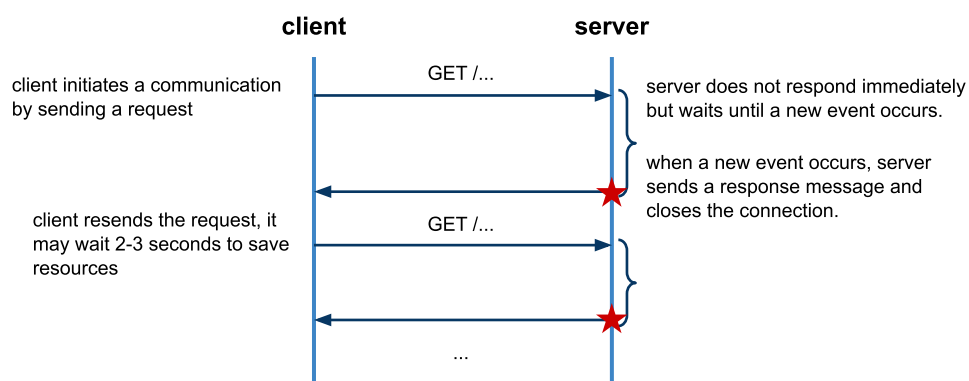
- Long-polling and Streaming
- WebSocket Protocol
- WebRTC

## Pushing and Polling



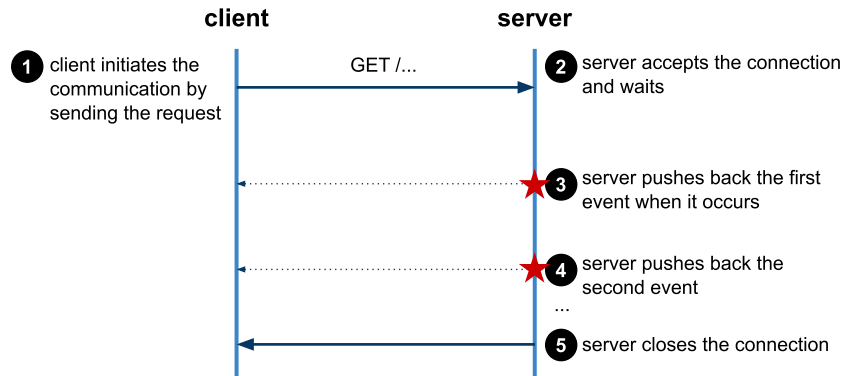
- Conceptual basis in messaging architectures
  - *event-driven architectures (EDA)*
- **HTTP is a request-response protocol**
  - *response cannot be sent without request*
  - *server cannot initiate the communication*
- **Polling** – client periodically checks for updates on the server
- **Pushing** – updates from the server (also called COMET)
  - = **long polling** – server holds the request for some time
  - = **streaming** – server sends updates without closing the socket

## HTTP Long Polling



- Server holds long-poll requests
  - *server responds when an event or a timeout occurs*
  - *saves computing resources at the server as well as network resources*
  - *can be applied over HTTP persistent and non-persistent communication*
- Issues:
  - *maximum time of the request processing at the server*
  - *concurrent requests processing at the server*

# HTTP Streaming



- server defers the response until an event or timeout is available
- when an event is available, server sends it back to client as part of the response; this does not terminate the connection
- server is able to send pieces of response w/o terminating the conn.
  - using **transfer-encoding** header in HTTP 1.1
  - using *End of File* in HTTP 1.0
  - (server omits **content-length** in the response)

## Chunked Response

- Transfer encoding **chunked**
  - It allows to send multiple sets of data over a single connection
  - a chunk represents data for the event

```
1 HTTP/1.1 200 OK
2 Content-Type: text/plain
3 Transfer-Encoding: chunked
4
5 25
6 This is the data in the first chunk
7
8 1C
9 and this is the second one
10
11 0
```

- Each chunk starts with hexadecimal value for length
- End of response is marked with the chunk length of 0

- Steps:
  - server sends HTTP headers and the first chunk (step 3)
  - server sends second and subsequent chunk of data (step 4)
  - server terminates the connection (step 5)

## Issues with Chunked Response

- Chunks vs. Events
  - *chunks cannot be considered as app messages (events)*
  - *intermediaries might "re-chunk" the message stream*
    - e.g., combining different chunks into a longer one
- Client Buffering
  - *clients may buffer all data chunks before they make the response available to the client application*
- HTTP streaming in browsers
  - *Server-sent events*

## Server-Sent Events

- W3C specification
  - *part of HTML5 specs, see Server-Sent Events [🔗](#)*
  - *API to handle HTTP streaming in browsers by using DOM events*
  - *transparent to underlying HTTP streaming mechanism*
    - *can use both chunked messages and EOF*
  - *same origin policy applies*
- **EventSource** interface
  - *event handlers: **onopen**, **onmessage**, **onerror***
  - *constructor **EventSource(url)** – creates and opens the stream*
  - *method **close()** – closes the connection*
  - *attribute **readyState***
    - **CONNECTING** – *The connection has not yet been established, or it was closed and the user agent is reconnecting.*
    - **OPEN** – *The user agent has an open connection and is dispatching events as it receives them.*
    - **CLOSED** – *The conn. is not open, the user agent is not reconnecting.*

## Example

- Initiating **EventSource**

```
1 | if (window.EventSource != null) {  
2 |   var source = new EventSource('your_event_stream.php');  
3 | } else {  
4 |   // Result to xhr polling :(  
5 | }
```

- Defining event handlers

```
1 | source.addEventListener('message', function(e) {  
2 |   // fires when new event occurs, e.data contains the event data  
3 | }, false);  
4 |  
5 | source.addEventListener('open', function(e) {  
6 |   // Connection was opened  
7 | }, false);  
8 |  
9 | source.addEventListener('error', function(e) {  
10 |   if (e.readyState == EventSource.CLOSED) {  
11 |     // Connection was closed  
12 |   }  
13 | }, false);
```

- *when the conn. is closed, the browser reconnects every ~3 seconds*  
→ can be changed using **retry** attribute in the message data

## Event Stream Format

- Format

- response's **content-type** must be **text/event-stream**
- every line starts with **data:**, event message terminates with 2 **\n** chars.
- every message may have associated **id** (is optional)

```
1 | id: 12345\n  
2 | data: first line\n  
3 | data: second line\n\n
```

- JSON data in multiple lines of the message

```
1 | data: {\n  
2 |   data: "msg": "hello world",\n  
3 |   data: "id": 12345\n  
4 | data: }\n\n
```

- Changing the reconnection time

- *default is 3 seconds*

```
1 | retry: 10000\n  
2 | data: hello world\n\n
```

# Server-side implementation

- Java Servlet

– *method doGet*

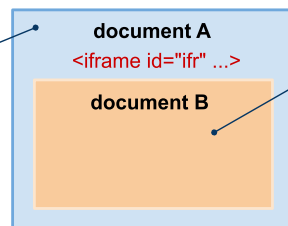
```
1 public void doGet(HttpServletRequest req, HttpServletResponse resp)
2     throws IOException {
3
4     // set http headers
5     resp.setContentType("text/event-stream");
6     resp.setHeader("cache-control", "no-cache");
7
8     // current time in milliseconds
9     long ms = System.currentTimeMillis();
10
11    // push data to the client for 20 seconds
12    // client should reconnect when the connection is closed
13    while (System.currentTimeMillis() - ms < 20000) {
14        resp.getWriter().print("data: servlet runs for " +
15                               (System.currentTimeMillis() - ms)/1000 + " seconds.\n\n");
16        resp.getWriter().flush();
17        try {
18            Thread.sleep(4000);
19        } catch (InterruptedException e) {
20            // do nothing;
21        }
22    }
23 }
```

# Other Technologies

- Cross-document messaging

script in document A

```
var o = document.getElementById("ifr");
o.contentWindow.postMessage("Hello world",
    "http://example.org/")
```



script in document B

```
window.addEventListener('message', receiver, false);
function receiver(e) {
    if (e.origin == 'http://example.com') {
        if (e.data == 'Hello world') {
            e.source.postMessage('Hello', e.origin);
        } else {
            alert(e.data);
        }
    }
}
```

– *The use of Cross Document Messaging for streaming*

1. The client loads a streaming resource in a hidden **iframe**
2. The server pushes a JavaScript code to the **iframe**
3. The browser executes the code as it arrives from the server
4. The embedded **iframe's** code posts a message to the upper document

## Overview

- Long-polling and Streaming
- **WebSocket Protocol**
- WebRTC

## WebSocket

- Specifications
  - IETF defines *WebSocket Protocol* [↗](#)
  - W3C defines *WebSocket API* [↗](#)
- Design principles
  - *a new protocol*
    - *browsers, web servers, and proxy servers need to support it*
  - *a layer on top of TCP*
  - *bi-directional communication between client and servers*
    - *low-latency apps without HTTP overhead*
  - *Web origin-based security model for browsers*
    - *same origin policy, cross-origin resource sharing*
  - *support multiple server-side endpoints*
- Two phases
  - *Handshake* – as an **upgrade** of a HTTP connection
  - *data transfer* – the protocol-specific on-the-wire data transfer

## Handshake – Request

- Request

- client sends a following HTTP request to upgrade the connection to WebSocket

```
1 GET /chat HTTP/1.1
2 Host: server.example.com
3 Upgrade: websocket
4 Connection: Upgrade
5 Sec-WebSocket-Key: dGh1IHhnbXBsZSBub25jZQ==
6 Sec-WebSocket-Origin: http://example.com
7 Sec-WebSocket-Protocol: chat, superchat
8 Sec-WebSocket-Version: 7
```

- **Connection** – request to upgrade the protocol
  - **Upgrade** – protocol to upgrade to
  - **Sec-WebSocket-Key** – a client key for later validation
  - **Sec-WebSocket-Origin** – origin of the request
  - **Sec-WebSocket-Protocol** – list of sub-protocols that client supports (proprietary)

## Handshake – Response

- Response

- server accepts the request and responds as follows

```
1 HTTP/1.1 101 Switching Protocols
2 Upgrade: websocket
3 Connection: Upgrade
4 Sec-WebSocket-Accept: s3pPLMBiTxaQ9kYGzzhZRbK+x0o=
5 Sec-WebSocket-Protocol: chat
```

- **101 Switching Protocols** – status code for a successful upgrade
  - **Sec-WebSocket-Protocol** – a sub-protocol that the server selected from the list of protocols in the request
  - **Sec-WebSocket-Accept** – a key to prove it has received a client WebSocket handshake request
  - Formula to compute **Sec-WebSocket-Accept**

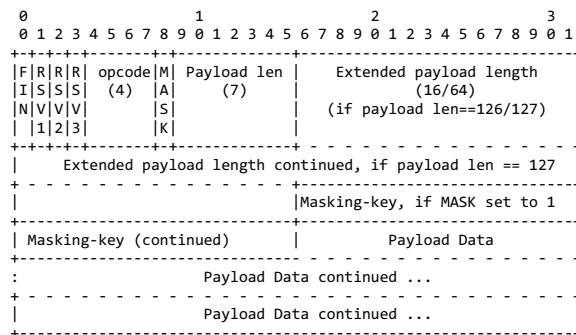
```
1 Sec-WebSocket-Accept = Base64Encode(SHA-1(Sec-WebSocket-Key +
2 "258EAF5-E914-47DA-95CA-C5AB0DC85B11"))
```

- **SHA-1** – hashing function
  - **Base64Encode** – Base64 encoding function
  - **"258EAF5-E914-47DA-95CA-C5AB0DC85B11"** – magic number



## Data Transfer

- After successful handshake
  - socket between the client and the "resource" at the server is established
  - client and the server can both read and write from/to the socket
  - No HTTP headers overhead
- Data Framing
  - Data transmitted in TCP packets (see RFC6455: Base Framing Protocol [↗](#))
  - Contains payload length, closing frame, ping, pong, type of data (text/binary), etc. and payload (message data)



## WebSocket API

- Client-side API
  - clients to utilize WebSocket, supported by Chrome, Safari
  - Hides complexity of WebSocket protocol for the developer
- JavaScript example

```

1 // ws is a new URL schema for WebSocket protocol; 'chat' is a sub-protocol
2 var connection = new WebSocket('ws://server.example.org/chat', 'chat');
3
4 // When the connection is open, send some data to the server
5 connection.onopen = function () {
6   // connection.protocol contains sub-protocol selected by the server
7   console.log('subprotocol is: ' + connection.protocol);
8   connection.send('data');
9 };
10
11 // Log errors
12 connection.onerror = function (error) {
13   console.log('WebSocket Error ' + error);
14 };
15
16 // Log messages from the server
17 connection.onmessage = function (e) {
18   console.log('Server: ' + e.data);
19 };
20
21 ...
22
23 // closes the connection
24 connection.close()

```

## Sockets.IO

- Many options for streaming
  - *long-polling, streaming, iframe, WebSockets*
  - *Not all browsers support WebSockets*
  - *Socket.IO* [↗](#) – *a layer providing a unified API*
- Sockets.IO
  - *API and JavaScript implementation*
  - *checks the availability of WebSocket protocol*
    - *fallback to long-polling or other technologies when not available*

```
1 // creates a new socket
2 var socket = new io.Socket();
3
4 // event handlers
5 socket.on('connect', function(){
6     socket.send('hi!');
7 })
8 socket.on('message', function(data){
9     alert(data);
10 })
11 socket.on('disconnect', function({}))
```

## Streaming video

- Webcams, IP or USB
  - *Play video stream using RTSP or M-JPEG*
  - *RTSP (Realtime Streaming Protocol) defines sequences to control palying multimedia*
- Sample tasks
  - *Add video stream to a web page*
    - **video** HTML5 element
  - *Capture frames from the camera and process them*
    - *Capture frames in a specific format such as JPG*
    - *Specific software to capture frames, typically OpenCV*
  - *Add annotation to video and expose as video stream*
    - *Detect objects in pictures – machine learning/deep learning*
    - *Mark objects and expose frames as video to the client*
    - *Create RTSP stream by using e.g. GStreamer or FFmpeg*
    - *Create stream of JPG images, so called M-JPEG and push them to the client*

## M-JPEG

- M-JPEG - Motion JPEG
  - Video compression format, each frame is represented as a JPEG image
  - Widely used by cameras today
  - Uses HTTP response stream of **multipart/x-mixed-replace** content type
- Example HTTP response to a M-JPEG request

```
1 HTTP/1.1 200 OK
2 Content-Type: multipart/x-mixed-replace; boundary=imgboundary
3
4 --imgboundary
5 Content-Type: image/jpeg
6 Content-length: 5432
7
8 [image 1 encoded jpeg data]
9
10 --imgboundary
11 Content-Type: image/jpeg
12 Content-length: 54335
13
14 [image 2 encoded jpeg data]
15
16 ...
17
```

## Overview

- Long-polling and Streaming
- WebSocket Protocol
- **WebRTC**

## WebRTC

- Web Real-Time Communication
  - *API to exchange media and arbitrary data between peers inside Web pages*
  - *It uses peer-to-peer principles*
  - *Supported by Google, Mozilla, Microsoft, Opera*
- Specifications
  - *WebRTC IETF Working Groups* [🔗](#)
  - *WebRTC W3C Working Groups* [🔗](#)
- History
  - *Google acquires company Global IP Solutions (GIPS) in 2010*
  - *GIPS developed underlying technology (codecs, echo cancellation techniques), released as open source*
  - *Google promoted the work around GIPS to W3C and IETF*

## WebRTC Main Tasks

- Acquiring audio and video
  - *JavaScript API: **MediaStream** (aka `getUserMedia`)*
- Communicating audio and video
  - *JavaScript API: **RTCPeerConnection***
- Communicating arbitrary data
  - *JavaScript API: **RTCDataChannel***

## getUserMedia

- JavaScript code

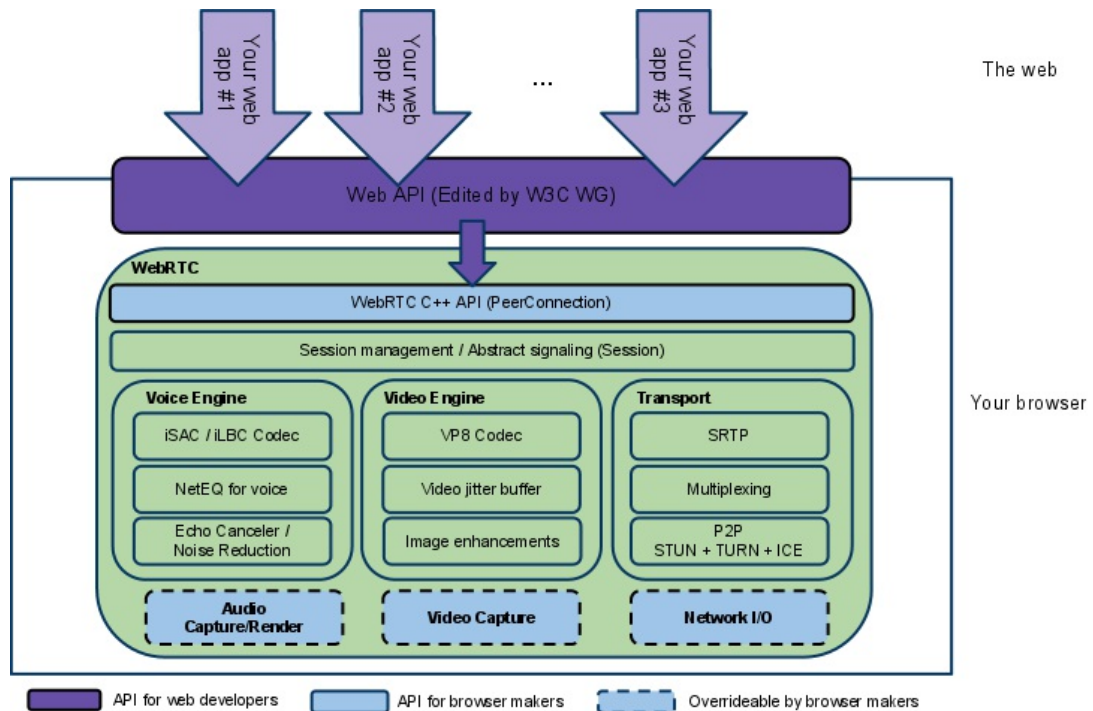
```
1  var constraints = {video: true};
2
3  function successCallback(stream) {
4      var video = document.querySelector("video");
5      video.src = window.URL.createObjectURL(stream);
6  }
7
8  function errorCallback(error) {
9      console.log("navigator.getUserMedia error: ", error);
10 }
11
12 navigator.getUserMedia(constraints, successCallback, errorCallback);
13
```

- Constraints
  - Control the contents of the *MediaStream*
  - Media type, resolution, frame rate
- JavaScript app can read and manipulate the stream.
- It is also possible to acquire **audio** as well as **screen capture**.

## RTCPeerConnection

- Allows to communicate media stream acquired by **getUserMedia**
  - Video chat, audio chat, screen sharing
- Some capabilities of **RTCPeerConnection**
  - Signal processing
  - Code handling
  - Peer to peer communication
  - Security
  - Bandwidth management

# WebRTC Architecture



## Communication

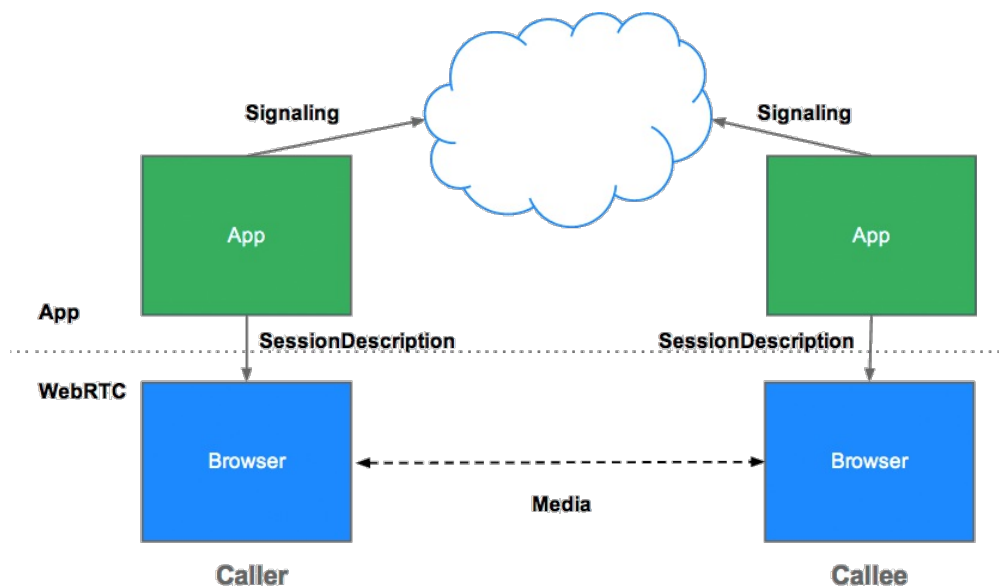
- Two phases
  1. **Signaling**
    - WebRTC defines **abstract signalling**
    - apps can use any signaling protocol, can use any such as SIP, XMPP, or custom using XHR or Websockets
  2. **Exchange of real-time data in peer-to-peer manner**
- Abstract signaling
  - Need to exchange **session description** objects
    - Formats, codecs the peers want to use
    - Network information for peer-to-peer communication
    - This information is captured as **RTCSessionDescription** (also SDP) structure
  - Any messaging mechanism and protocol

# SIP and SDP

- Standards
  - SIP – Session Initiation Protocol, protocol to establish and modify sessions.
  - SDP – Session Description Protocol, describes media for a session, defined in RFC4566 - Session Description Protocol [🔗](#)
- SDP Example

```
v=0
o=- 7614219274584779017 2 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE audio video
a=msid-semantic: WMS
m=audio 1 RTP/SAVPF 111 103 104 0 8 107 106 105 13 126
c=IN IP4 0.0.0.0
a=rtcp:1 IN IP4 0.0.0.0
a=ice-ufrag:W2TGCZw2NZHuwlnf
a=ice-pwd:xdQEccP40E+P0L5qTyzDgfmW
a=extmap:1 urn:iETF:params:rtp-hdext:ssrc-audio-level
a=mid:audio
a=rtcp-mux
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:9c1AHZ27dZ9xPI91YNfS1I67/EMkjHHIHORiClQe
a=rtpmap:111 opus/48000/2
...
```

# Signaling Diagram



## JavaScript Session Establishment (JSEP)

- JSEP is a protocol to create a session between two parties
  - *The interface needed by an application to deal with the negotiated local and remote session descriptions*
- JSEP steps between Alice and Bob
  1. Alice creates an offer that contains her local SDP.
  2. Alice attaches that offer to `RTCPeerConnection` object.
  3. Alice sends the offer to a signaling server using custom-built mechanism (`WebSocket`, `XHR`, etc.)
  4. Bob receives Alice's offer from the signaling server
  5. Bob creates an answer using his local SDP.
  6. Bob attaches his answer along with Alice's offer to his own `RTCPeerConnection` object.
  7. Bob returns his answer to the signaling server.
  8. Alice receives Bob's offer from the signaling server.

## Interactive Connectivity Establishment

- ICE – Interactive Connectivity Establishment
  - *Allows WebRTC to overcome complexities of real-world networking*
  - *Finds the best path to connect peers such as*
    - *direct P2P communication.*
    - *by using STUN or TURN servers.*
- STUN – Session Traversal Utilities for NAT
  - *Allows to discover the presence of a NAT server:*
  - *Allows to discover the public IP address and a port that the NAT has allocated for UDP flows.*
  - *It is provided as a third-party network server (STUN server) located on the public side of the NAT.*
- TURN – Traversal Using Relays around NAT
  - *Communication relay for hosts behind NAT when STUN does not work.*



