

Web 2.0

Lecture 7: Protocols for the Realtime Web

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Evropský sociální fond
Praha & EU: Investujeme do vaší budoucnosti

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Overview

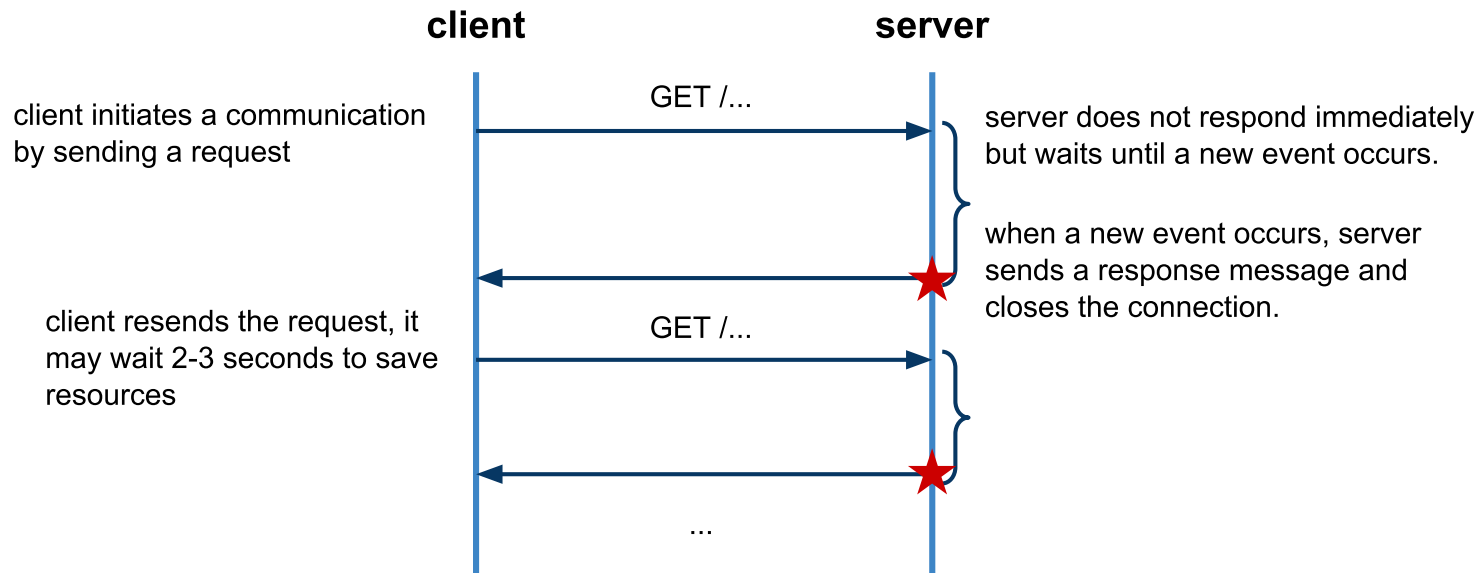
- Long-polling and Streaming
- WebSocket Protocol
- WebRTC

Pushing and Polling



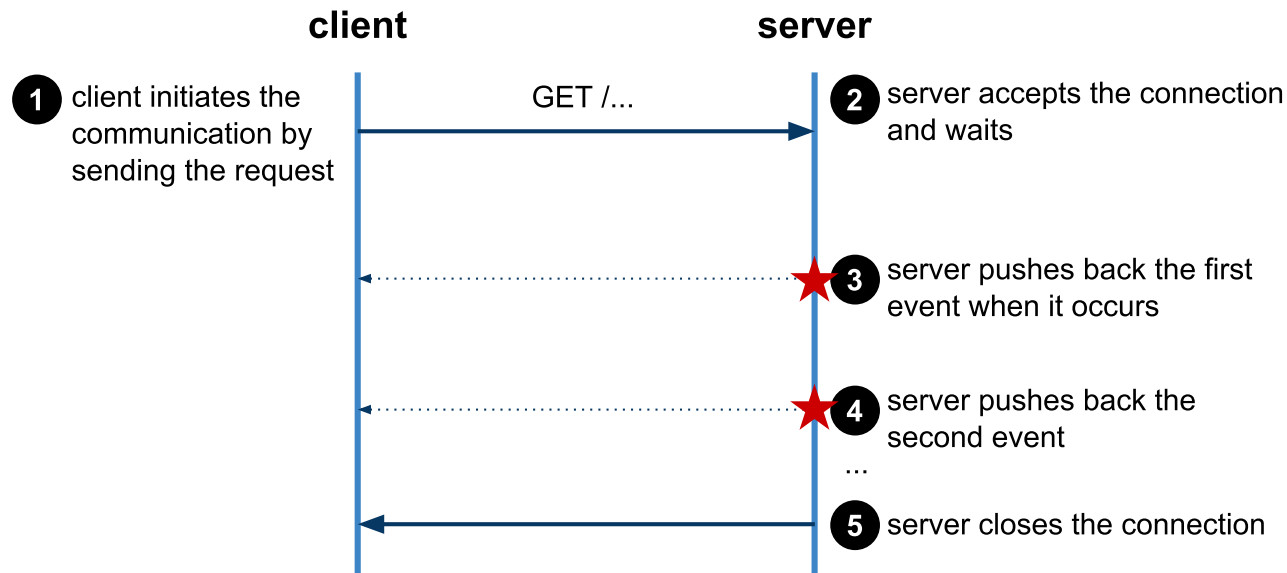
- Conceptual basis in messaging architectures
 - *event-driven architectures (EDA)*
- **HTTP is a request-response protocol**
 - *response cannot be sent without request*
 - *server cannot initiate the communication*
- **Polling** – client periodically checks for updates on the server
- **Pushing** – 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`

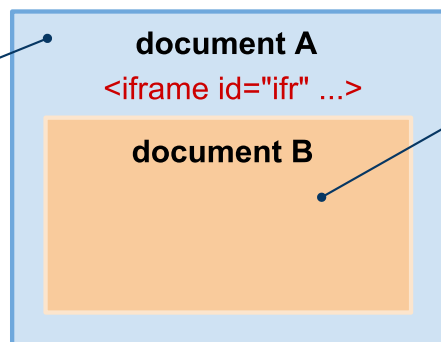
```
1  public void doGet(HttpServletRequest req, HttpServletResponse resp)
2      throws IOException {
3
4      // set http headers
5      resp.setContentType("text/event-stream");
6      resp.setHeader("cache-control", "no-cache");
7
8      // current time in milliseconds
9      long ms = System.currentTimeMillis();
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*

- Channel API

- *Google Technology for streaming API for AppEngine*
- *not based on HTTP streaming*
- *utilizes XMPP capabilities + hidden iframe at client-side*

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: dGhlIHNhbXBsZSBub25jZQ==
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 | "258EAF5E-E914-47DA-95CA-C5AB0DC85B11"))
```

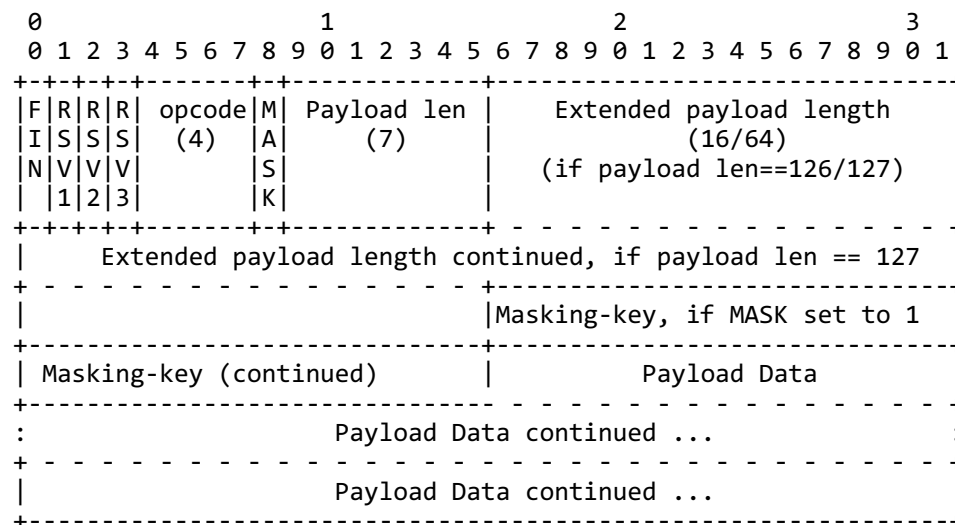
- **SHA-1** – *hashing function*

- **Base64Encode** – *Base64 encoding function*

- **"258EAF5E-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(){}))
```

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

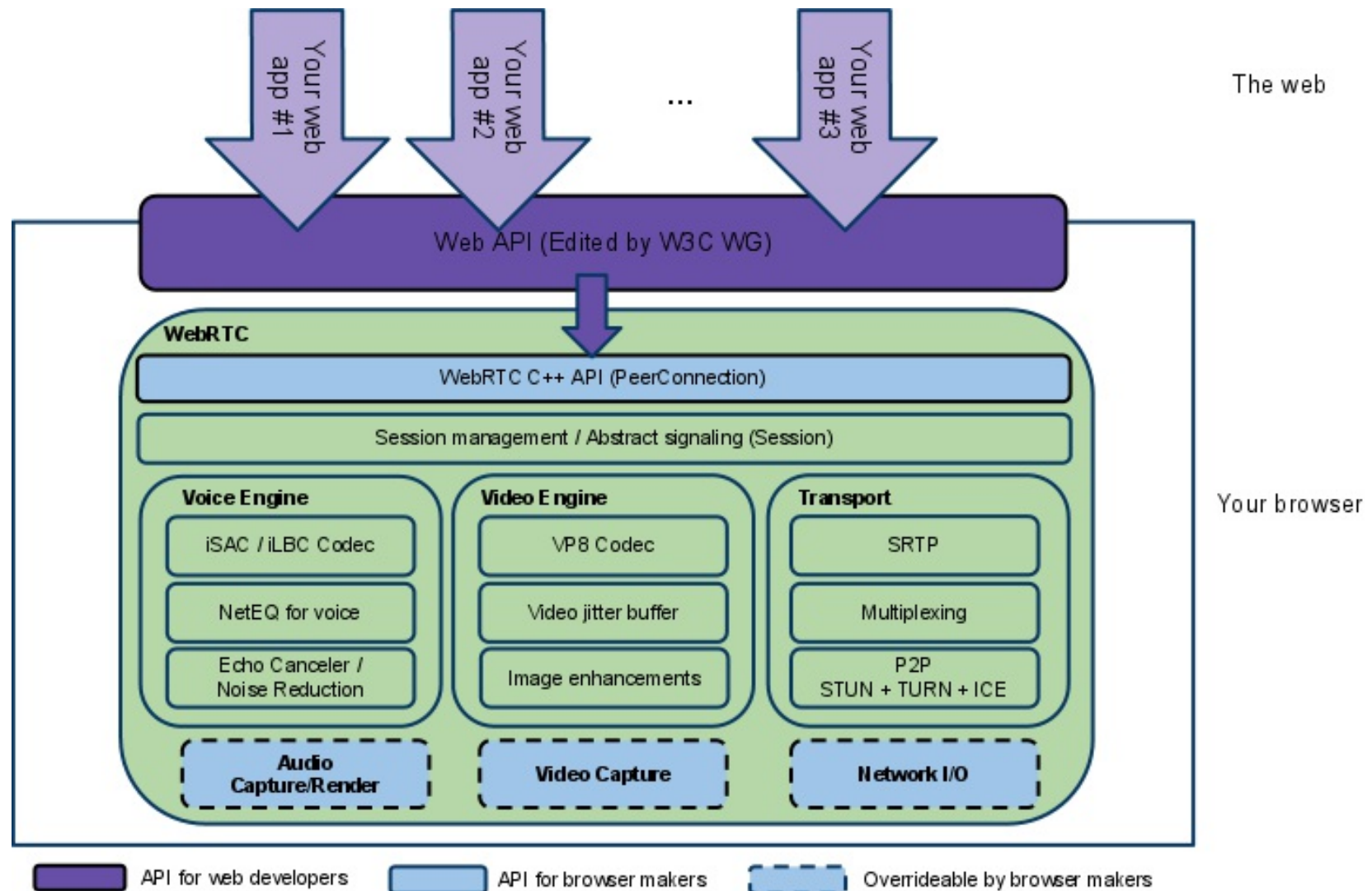
- *Control the contents of the MediaStream*
- *Media type, resolution, frame rate*

- 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

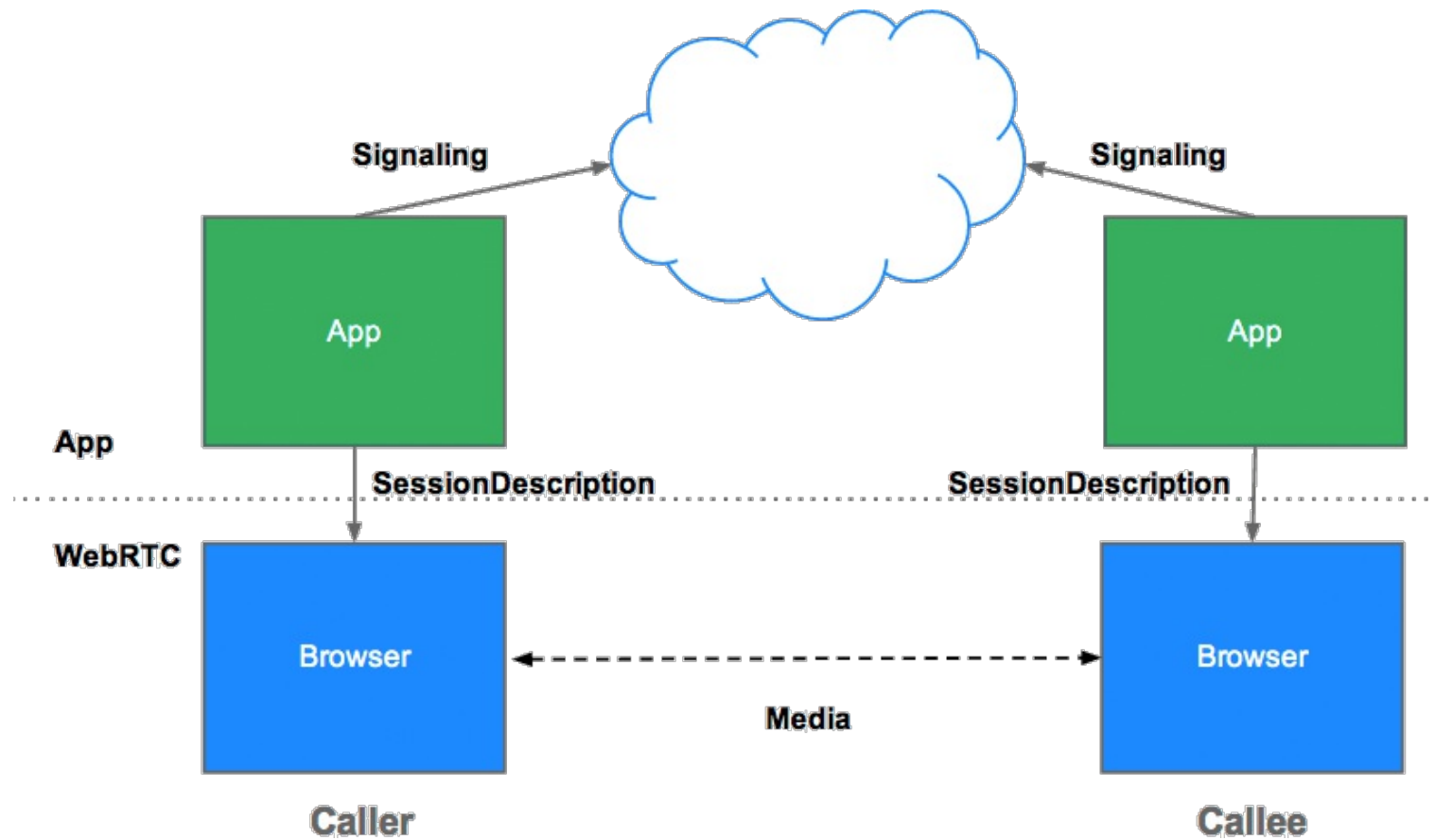
- Two phases
 1. *Signaling*
 - WebRTC defines **abstract signalling**
 - apps can use any signaling protocol, can use any such as SIP, XMPP, or custom using XHR or Websockets
 2. *Exchange of real-time data in peer-to-peer manner*
- Abstract signaling
 - Need to exchange **session description** objects
 - 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-hdrext:ssrc-audio-level
a=mid:audio
a=rtcp-mux
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:9c1AHz27dZ9xPI91YNfSlI67/EMkjHHIHORiClQe
a=rtpmap:111 opus/48000/2
...
```

Signaling Diagram



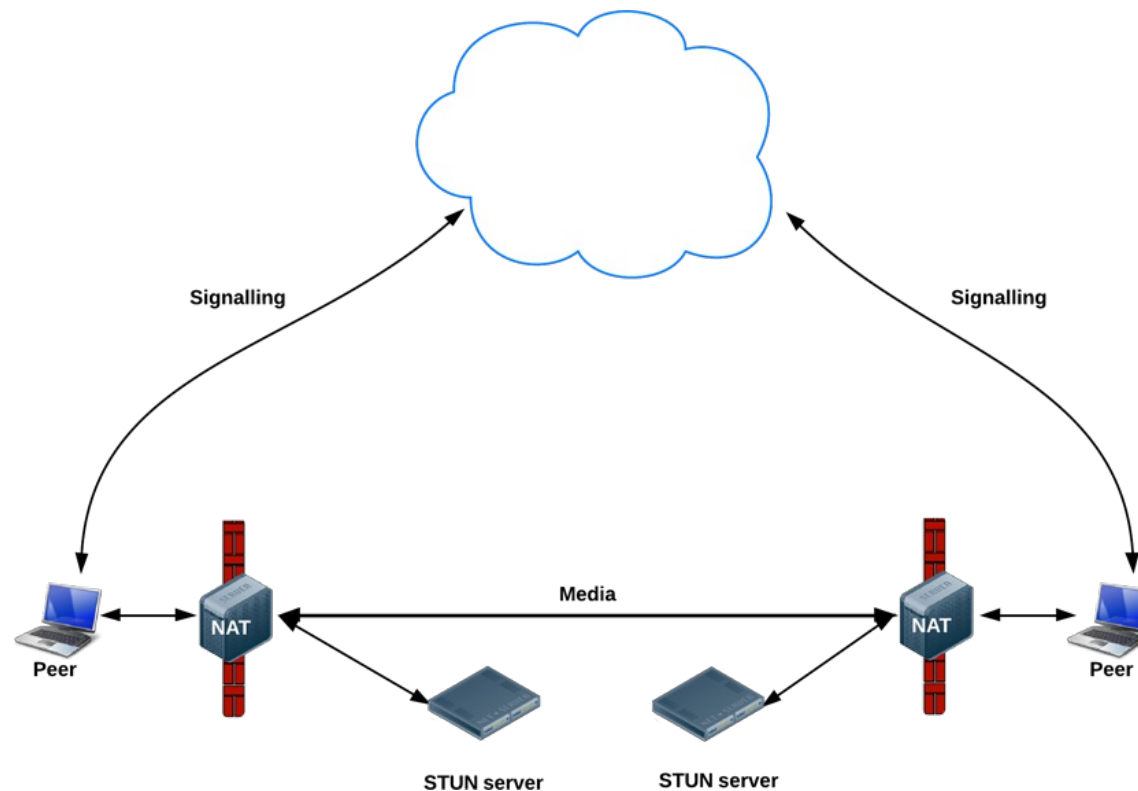
JavaScript Session Establishment (JSEP)

- JSEP is a protocol to create a session between two parties
- 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

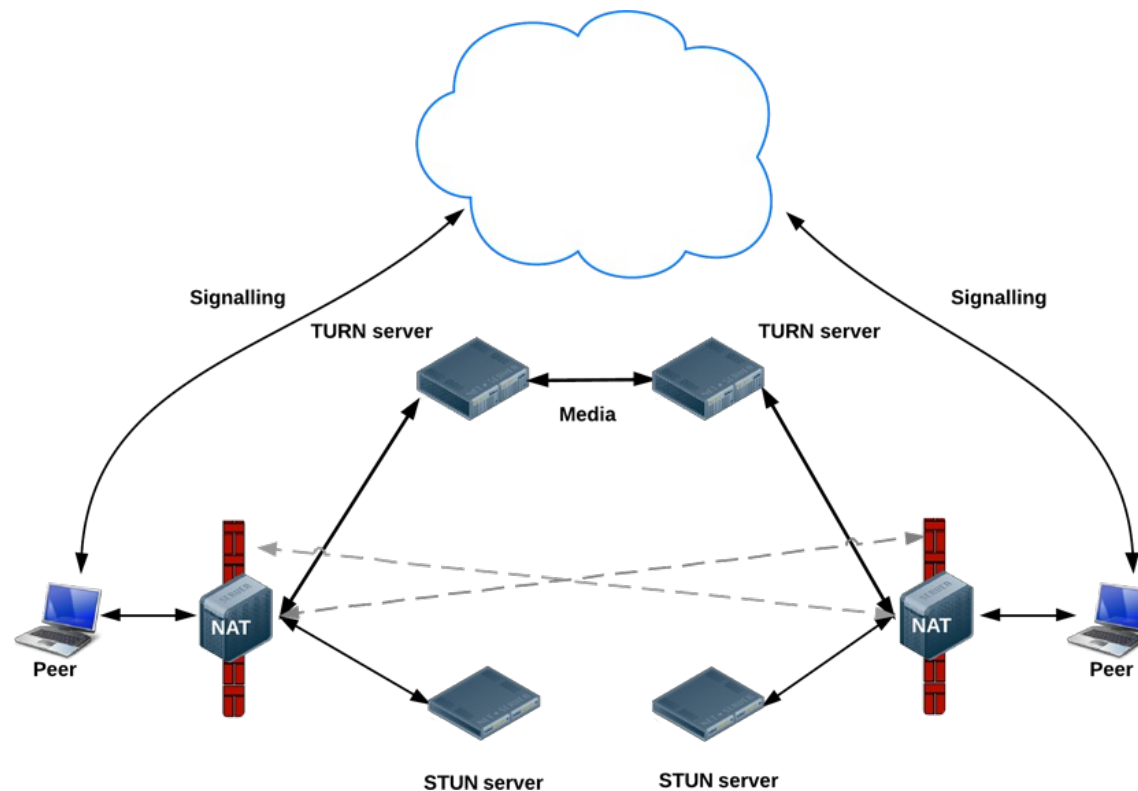
- ICE – Interactive Connectivity Establishment
 - *Allows WebRTC to overcome complexities of real-world networking*
 - *Finds the best path to connect peers such as*
 - *direct P2P communication.*
 - *by using STUN or TURN servers.*
- STUN – Session Traversal Utilities for NAT
 - *Allows to discover 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.*

STUN



- *STUN is a simple server, cheap to run*
- *Data flows peer-to-peer*

TURN



- a cloud fallback when peer-to-peer does not work
- data sent via a relay server, uses server bandwidth