

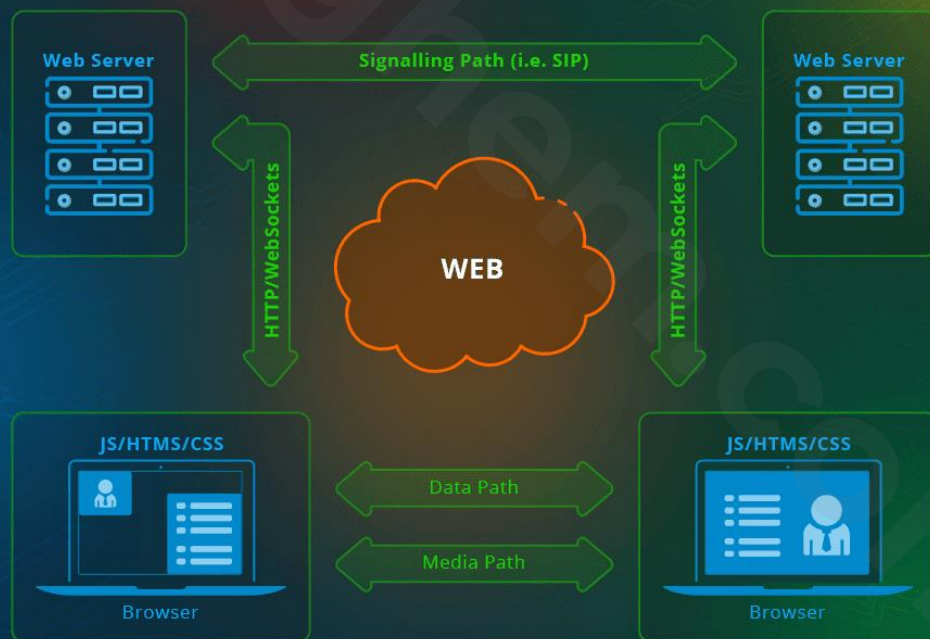
# TELEPHONY OVER IP SOLUTION

## Travaux pratiques – 2

### WebRTC

Classe : E4CCSN

How does WebRTC work?

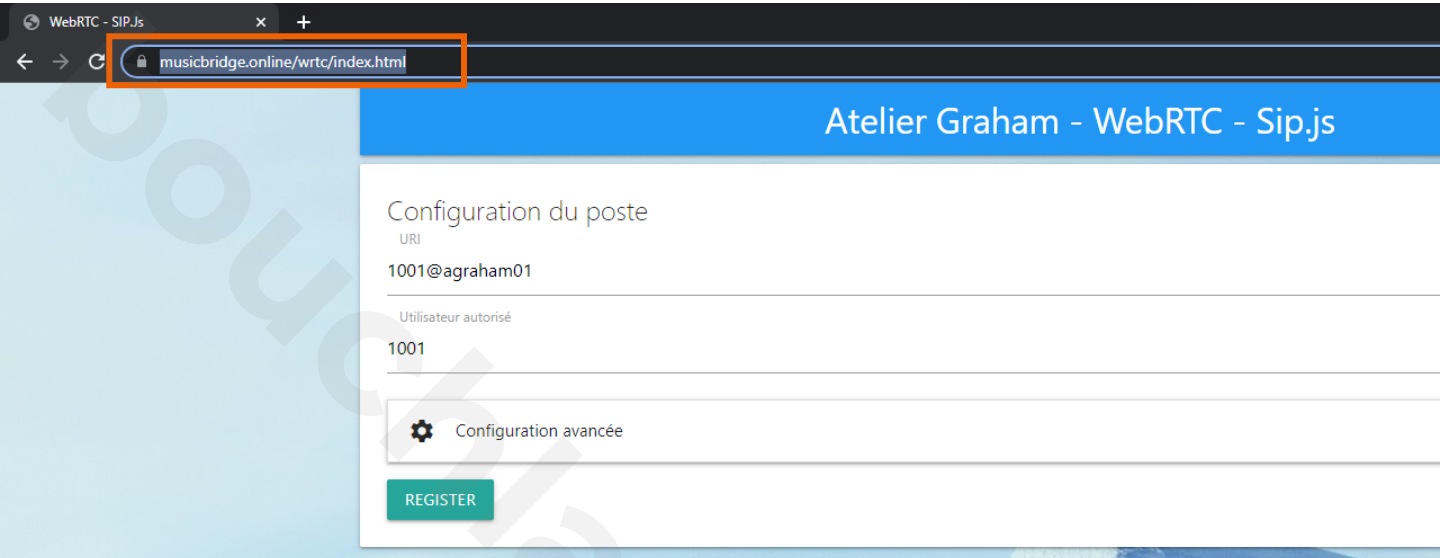


Réalisé par : Mohammed BOUCLAGHEM

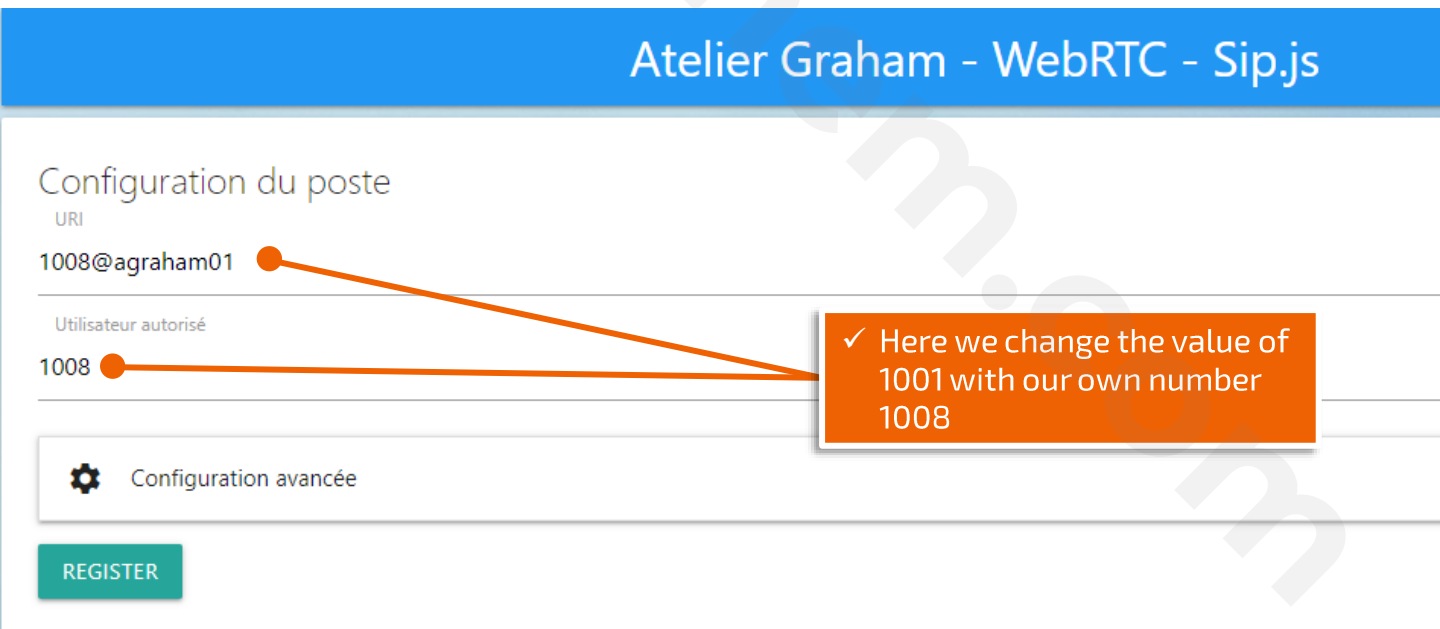
# 1) launching and configuration :

## 1.1 Lunching Firefox and browsing this link:

<https://musicbridge.online/wrtc/index.html>



## 1.2 Login using our account username :



# 1) launching and configuration :

## 1.3 First look into the WebRTC Console :

The screenshot shows the 'WebRTC Console' interface. At the top, there's a blue header with the text 'Atelier Graham - WebRTC - Sip.js'. Below the header, there's a form with a label 'Numéro à appeler' and the number '1002' entered. To the right of the form are two buttons: a red one labeled 'DÉCONNEXION' and a green one labeled 'APPELER'. Below the form are two buttons: 'Transfer' and 'Novideo'. At the bottom, there's a section titled 'Evenements' containing two entries: 'Poste 1010@agraham01 enregistré' and 'Poste 1008@agraham01 enregistré'. Two orange callout boxes with checkmarks point to the 'Numéro à appeler' input and the 'Evenements' list. The first callout says '✓ In this input we can chose who we call with their number'. The second callout says '✓ Here we can see all the events'.

## 1.4 To access to the JavaScript Console we press : **Ctrl** + **Shift** + **J**

This screenshot shows a browser window with the WebRTC Console interface and the JavaScript Console open. The WebRTC Console is the same as in the previous screenshot. The JavaScript Console is open at the bottom, showing a log of events. The first event is a 'message-summary' for a SIP message. The message details are: 'Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE', 'Event: message-summary', 'Allow-Events: talk, hold, conference, presence, as-feature-event, dialog, line-seize, call-info, sla, include-session-description, presence.winfo, message-summary, refer', 'Subscription-State: terminated;reason=noresource', 'Content-Type: application/simple-message-summary', 'Content-Length: 61', 'Messages-Waiting: no', 'Message-Account: sip:1008@agraham01'. Below this, there's a timestamp and the text 'sending WebSocket message:'. The console also shows a SIP error: 'SIP/2.0 481 Call/Transaction Does Not Exist', 'Via: SIP/2.0/MS 194.5.159.151:7443;port=branch-296483097kpv8a958Q', 'From: <sip:1008@194.5.159.151>;tag=9726402155e', 'To: <sip:1008@194.5.159.151>;tag=qcm453r111', 'CSeq: 63734764 NOTIFY', 'Call-ID: 33e37145-2b06-123c-2cad-4f7901eb0037', 'Supported: outbound', 'User-Agent: SIP.js/0.15.1', 'Content-Length: 0'. At the bottom of the browser window, there are some updates from Chrome 109, including 'Recorder panel updates' and 'Improved JavaScript debugging'.

## 2) Observation of the console :

```
Elements Console Sources Network Performance Memory Application Security Lighthouse Recorder Performance insights
top Filter
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | The UA class has been deprecated and will no longer be available starting with SIP.js release 0.16.0. The UA has been replaced by the UserAgent class. Please update accordingly.
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | configuration parameters after validation:
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - viaHost: "192.0.2.161"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - uri: sip:1008@agraham01
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - custom: {}
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - displayName: ""
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - password: NOT SHOWN
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - register: true
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - registerOptions: {}
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - transportConstructor: Transport
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - transportOptions: {"wsServers":["wss://musicbridge.online:7443"],"trace":
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - usePreloadedRoute: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - userAgentString: "SIP.js/0.15.1"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - noAnswerTimeout: 60000
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - hackViaTcp: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - hackViaContact: true
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - hackViaTransport: true
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - hackAllowRegisteredOptionTags: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - sessionDescriptionHandlerFactoryOptions: {"constraints":{},"peerConnection
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - extraSupported: []
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - contactName: "2sp5utr"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - contactTransport: "wss"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - forceRport: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - autostart: true
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - autostop: true
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - rell00: "none"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - dtmfType: "info"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - replaces: "none"
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - sessionDescriptionHandlerFactory: function (session, options) {
  var logger = (session instanceof session.Session) ?
    session.userAgent.getLogger("sip.sessionDescriptionHandler", session.id) :
    session.ua.getLogger("sip.inviteContext.sessionDescriptionHandler", session.id);
  var observer = new SessionDescriptionHandlerObserver_1.SessionDescriptionHandlerObserver(session, options);
  return new SessionDescriptionHandler(logger, observer, options);
}
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - authenticationFactory: undefined
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - allowLegacyNotifications: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - allowOutOfDialogRefers: false
Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.ua | - experimentalFeatures: false
```

✓ We can see on the console the events when we register or we login using the username and the other configurations options with time and the date

```
DevTools - musicbridge.online/wrtc/index.html
Elements Console Sources Network Performance Memory Application Security Lighthouse Recorder Performance insights
top Filter
REGISTER sip:agraham01 SIP/2.0
Via: SIP/2.0/WSS 192.0.2.161;branch=z9hG4bK5342267
To: <sip:1008@agraham01>
From: <sip:1008@agraham01>;tag=djqbrj35r6
CSeq: 9747 REGISTER
Call-ID: jdppipa4ci1l01jrt7hje
Max-Forwards: 70
Contact: <sip:2sp5utrr@192.0.2.161;transport=wss>;expires=600
Allow: ACK,CANCEL,INVITE,MESSAGE,BYE,OPTIONS,INFO,NOTIFY,REFER
Supported: outbound, path, gruu
User-Agent: SIP.js/0.15.1
Content-Length: 0

Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.transport | received WebSocket text message:

SIP/2.0 401 Unauthorized
Via: SIP/2.0/WSS 192.0.2.161;branch=z9hG4bK5342267;received=91.174.236.160;rport=32791
From: <sip:1008@agraham01>;tag=djqbrj35r6
To: <sip:1008@agraham01>;tag=H2SaQr6Up1Q0m
Call-ID: jdppipa4ci1l01jrt7hje
CSeq: 9747 REGISTER
User-Agent: FreeSWITCH-mod_sofia/1.10.8-release.14-64bit
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
Supported: timer, path, replaces
WWW-Authenticate: Digest realm="agraham01", nonce="d8798120-af85-4447-8df4-4b1f3e683dbe", algorithm=MD5, qop="auth"
Content-Length: 0

Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.transport | sending WebSocket

REGISTER sip:agraham01 SIP/2.0
Via: SIP/2.0/WSS 192.0.2.161;branch=z9hG4bK6499500
To: <sip:1008@agraham01>
From: <sip:1008@agraham01>;tag=djqbrj35r6
CSeq: 9748 REGISTER
Call-ID: jdppipa4ci1l01jrt7hje
Max-Forwards: 70
Authorization: Digest algorithm=MD5, username="1008", realm="agraham01", nonce="d8798120-af85-4447-8df4-4b1f3e683dbe", uri="sip:agraham01", response="e272952
Contact: <sip:2sp5utrr@192.0.2.161;transport=wss>;expires=600
Allow: ACK,CANCEL,INVITE,MESSAGE,BYE,OPTIONS,INFO,NOTIFY,REFER
Supported: outbound, path, gruu
User-Agent: SIP.js/0.15.1
Content-Length: 0

Thu Feb 16 2023 21:06:05 GMT+0100 (Central European Standard Time) | sip.transport | received WebSocket text message:

SIP/2.0 200 OK
Via: SIP/2.0/WSS 192.0.2.161;branch=z9hG4bK6499500;received=91.174.236.160;rport=32791
From: <sip:1008@agraham01>;tag=djqbrj35r6
```

✓ And If we scroll down we can see the Register request and the login information (challenge)...

✓ And here we get the authorization for the username 1008

## 2) Observation of the source codes :

HTML File

```
<!-- Answer / Refuse Call -->
<div class="row">
  <a id="yes_answer_btn" class="waves-effect waves-light btn disabled green col s5 hide">Accepter appel</a>
  <div class="col s2"></div>
  <a id="no_answer_btn" class="waves-effect waves-light btn disabled red col s5 hide">Refuser appel</a>
</div>

<!-- Hang Up Call -->
<div id="hangupRow" class="row hide">
  <a id="hangup_btn" class="waves-effect waves-light btn red col s4 offset-s4">Raccrocher</a>
</div>

<!-- Audio/Video elements for calls -->
<div class="row">
  <video id="remoteVideo" class="show"></video>
  <div class="col s2"></div>
  <video id="localVideo" class="show"></video>
</div>

<!-- Global logs -->
<ul class="collapsible">
  <li class="active">
    <div class="collapsible-header"><i class="material-icons">textsms</i>Evenements</div>
    <div class="collapsible-body">
      <div id="phone_log" style="max-height:200px;overflow:auto;">
      </div>
    </div>
  </li>
</ul>

<!-- Log SIP -->
<ul class="collapsible">
  <li>
    <div class="collapsible-header"><i class="material-icons">textsms</i>Log SIP</div>
    <div class="collapsible-body">
      <div id="log_display" style="max-height:200px;overflow:auto;">
      </div>
    </div>
  </li>
</ul>

</div>
</div>
</div>

<!-- Include JS script -->
<script src="./renderer.js"></script>
<script src="./sip-0.15.1.js"></script>
</body>
</html>
```

- ✓ Here we have the source code :
- ✓ **HTML (index.html)** : the html code<sup>1</sup> that implements the difference options and interactions
- ✓ And in the HTML file there is a part of that include a **JavaScript** file source : (`./sip-0-15.1.js`)<sup>2</sup>

```
1327 parse the given header on the given index.
1328 * @param name - header name
1329 * @param idx - header index
1330 * @returns Parsed header object, undefined if the
1331 * header is not present or in case of a parsing error.
1332 */
1333 IncomingMessage.prototype.parseHeader = function (name, idx) {
1334   if (idx === void 0) { idx = 0; }
1335   name = utils_1.headerize(name);
1336   if (!this.headers[name]) {
1337     // this.logger.log("header '" + name + "' not present");
1338     return;
1339   }
1340   else if (idx >= this.headers[name].length) {
1341     // this.logger.log("not so many '" + name + "' headers present");
1342     return;
1343   }
1344   var header = this.headers[name][idx];
1345   var value = header.raw;
1346   if (header.parsed) {
1347     return header.parsed;
1348   }
1349   // substitute '-' by '_' for grammar rule matching.
1350   var parsed = grammar_1.Grammar.parse(value, name.replace(/-/g, "_"));
1351   if (parsed === -1) {
1352     this.headers[name].splice(idx, 1); // delete from headers
1353     // this.logger.warn('error parsing "' + name + "' header from www.bouchlaghem.com');
1354     return;
1355   }
1356   return header.parsed;
1357 }
```

- ✓ The JavaScript file contain all the functions that make the website dynamic and interactive

# 3 ) Call a registered colleague :

## Atelier Graham - WebRTC - Sip.js

WebRTC Console

DÉCONNECTION

Numéro à appeler

APPELER

1011

Transfer Novideo

RACCROCHER

Evenements

Début de l'appel vers 1011

PROGRESS

PROGRESS

Le destinataire a accepté la communication, liaison en cours

✓ Here we make a call for a registered colleague, and as we can see its works just fine let us now check the logs under

DevTools - musicbridge.online/wrtc/index.html

Elements Console Sources Network Performance Memory Application Security Lighthouse Recorder Performance insights

top Filter

Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.transport | received WebSocket text message:

```
SIP/2.0 200 OK
Via: SIP/2.0/WSS 192.0.2.124;branch=z9hG4bK7316876;received=91.174.236.160;rport=38447
From: <sip:1008@graham01>;tag=nrv199k44e
To: <sip:1011@graham01>;tag=6SQ40y7UNHK3D
Call-ID: m8d1mthmtkinilte12e
CSeq: 2 INVITE
Contact: <sip:1011@194.5.159.151:5060>;transport=udp
User-Agent: FreeSWITCH-mod_sofia/1.10.0-release-14-64bit
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
Supported: timer, path, replaces
Allow-Events: talk, hold, conference, presence, as-feature-event, dialog, line-seize, call-info, sla, include-session-description, presence.winfo, message-summary, refer
Content-Type: application/sdp
Content-Disposition: session
Content-Length: 892
Remote-Party-ID: "Outbound Call" <sip:3gvub37@graham01>;party=calling;privacy=off;screen=no
```

```
v=0
o=FreeSWITCH 1676558025 1676558026 IN IP4 194.5.159.151
s=FreeSWITCH
c=IN IP4 194.5.159.151
t=0
a=msid-semantic: WMS DA18Q5zNpZM4N1FKXNfL6FqYwORnkeR
m=audio 22810 UDP/TLS/RTP/SAVPF 111 110
a=rtpmap:111 opus/48000/2
a=freq:111 useinbandfec=1; minptime=10
a=rtpmap:110 telephone-event/48000
a=ptime:20
a=fingerprint:sha-256 98:BA:1C:40:5E:A2:3C:1D:DB:28:EB:85:B6:24:3F:87:FD:A0:1C:80:85:C9:8B:31:C2:88:E9:9D:27:5D:F3:A6
a=setup:active
a=rtp-mux
a=rtp:22810 IN IP4 194.5.159.151
a=ice-frag:qohUxLJqr7AV0b
a=ice-pwd:lTtYnWtU4IDHK9YzGokDNjP
a=candidate:9837866711 1 udp 2130706431 194.5.159.151 22810 typ host generation 0
a=end-of-candidates
a=ssrc:1207823003 cname:KpPr7f4n461SNTF
a=ssrc:1207823003 msid:DA18Q5zNpZM4N1FKXNfL6FqYwORnkeR a0
a=ssrc:1207823003 mslabel:DA18Q5zNpZM4N1FKXNfL6FqYwORnkeR
a=ssrc:1207823003 label:DA18Q5zNpZM4N1FKXNfL6FqYwORnkeRa0
```

✓ Here we see that the call start via the SIP/2.0/WSS 192.0.124 From 1008 to 1011 and with all the configuration and the codecs that used on the call

```
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.invitecontext.sessionDescriptionHandler | m8d1mthmtkinilte12enrv199k44e | ICE Connection State changed to iceConnectionChecking
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.invitecontext.sessionDescriptionHandler | m8d1mthmtkinilte12enrv199k44e | track added
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.invite-dialog | INVITE dialog m8d1mthmtkinilte12enrv199k44e6SQ40y7UNHK3D sending ACK request
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.transport | sending WebSocket message
```

```
ACK sip:1011@194.5.159.151:5060;transport=udp SIP/2.0
Via: SIP/2.0/WSS 192.0.2.124;branch=z9hG4bK9835296
To: <sip:1011@graham01>;tag=6SQ40y7UNHK3D
From: <sip:1008@graham01>;tag=nrv199k44e
CSeq: 2 ACK
Call-ID: m8d1mthmtkinilte12e
Max-Forwards: 70
Supported: outbound
User-Agent: SIP.js/0.15.1
Content-Length: 0
```

✓ The exchange between the server and the client the ICE Connection State and the ACK request with the sent of the WebSocket message

```
Uncaught (in promise) DOMException: The play() request was interrupted by a new load request. https://goo.gl/LdLk22
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) | sip.invitecontext.sessionDescriptionHandler | m8d1mthmtkinilte12enrv199k44e | ICE Connection State changed to iceConnectionConnected
```



### 3 ) Call a registered colleague :

```
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | waitforIceGatheringComplete | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | ICE is not complete. Returning promise | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | RTCCIceGatheringState changed: gathering | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | ICE candidate received: candidate:4006265929 1 udp 2113937151 80cb196f-ac24-47b2-8bec-910c3e60ce04.local 56400 | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | ICE candidate received: candidate:1835802825 1 udp 1677729535 91.174.236.160 40241 typ srflx raddr 0.0.0.0 rport | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | RTCCIceGatheringState changed: complete | sip:0.15.1.js:18080
Thu Feb 16 2023 21:53:54 GMT+0100 (Central European Standard Time) sip.transport | sending WebSocket message: | sip:0.15.1.js:18080
```

✓ Here we can see the ICE on details from waiting for the gathering to complete, here our machine share the addresses, and we have just one candidate the server

```
INVITE sip:1011@graham01 SIP/2.0
Via: SIP/2.0/MSS 192.0.2.124;branch=296648K121120
To: <sip:1011@graham01>
From: <sip:1008@graham01>;tag=rv199k44e
CSeq: 1 INVITE
Call-ID: m8dl1mthtkin1tel2e
Max-Forwards: 70
Contact: <sip:3nkqu0@192.0.2.124>;transport=ws;ob
Allow: ACK,CANCEL,INVITE,MESSAGE,BYE,OPTIONS,INFO,NOTIFY,REFER
Supported: outbound
User-Agent: SIP.js/0.15.1
Content-Type: application/sdp
Content-Length: 1737
```

```
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | ICE Connection State changed to iceConnectionChecking | sip:0.15.1.js:18080
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | track added | sip:0.15.1.js:18080
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.invite-dialog | INVITE dialog m8dl1mthtkin1tel2enrv199k44e65Q4b/7UNHk3D sending ACK request | sip:0.15.1.js:18080
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.transport | sending WebSocket message: | sip:0.15.1.js:18080
```

```
ACK sip:1011@194.5.159.151:5860;transport=udp SIP/2.0
Via: SIP/2.0/MSS 192.0.2.124;branch=296648K9835296
To: <sip:1011@graham01>;tag=65Q4b/7UNHk3D
From: <sip:1008@graham01>;tag=rv199k44e
CSeq: 2 ACK
Call-ID: m8dl1mthtkin1tel2e
Max-Forwards: 70
Supported: outbound
User-Agent: SIP.js/0.15.1
Content-Length: 0
```

```
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | ICE Connection State changed to iceConnectionConnected | sip:0.15.1.js:18080
Thu Feb 16 2023 21:54:19 GMT+0100 (Central European Standard Time) sip.transport | received WebSocket text message: | sip:0.15.1.js:18080
```

```
BYE sip:3nkqu0@192.0.2.124;transport=ws;ob SIP/2.0
Via: SIP/2.0/MSS 194.5.159.151:7443;rport;branch=296648K8p6eQpSfckrhj
Max-Forwards: 70
From: <sip:1011@graham01>;tag=65Q4b/7UNHk3D
To: <sip:1008@graham01>;tag=rv199k44e
Call-ID: m8dl1mthtkin1tel2e
CSeq: 63736720 BYE
User-Agent: FreeSWITCH-mod_sofia/1.10.8-release-14-64bit
Allow: INVITE, ACK, BYE, CANCEL, OPTIONS, MESSAGE, INFO, UPDATE, REGISTER, REFER, NOTIFY, PUBLISH, SUBSCRIBE
Supported: timer, path, replaces
Reason: Q.850;cause=16;text="NORMAL_CLEARING"
Content-Length: 0
```

```
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.invite-dialog | INVITE dialog m8dl1mthtkin1tel2enrv199k44e65Q4b/7UNHk3D received BYE request | sip:0.15.1.js:18080
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.transport | sending WebSocket message: | sip:0.15.1.js:18080
```

```
SIP/2.0 200 OK
Via: SIP/2.0/MSS 194.5.159.151:7443;rport;branch=296648K8p6eQpSfckrhj
From: <sip:1011@graham01>;tag=65Q4b/7UNHk3D
To: <sip:1008@graham01>;tag=rv199k44e
CSeq: 63736720 BYE
Call-ID: m8dl1mthtkin1tel2e
Supported: outbound
User-Agent: SIP.js/0.15.1
Content-Length: 0
```

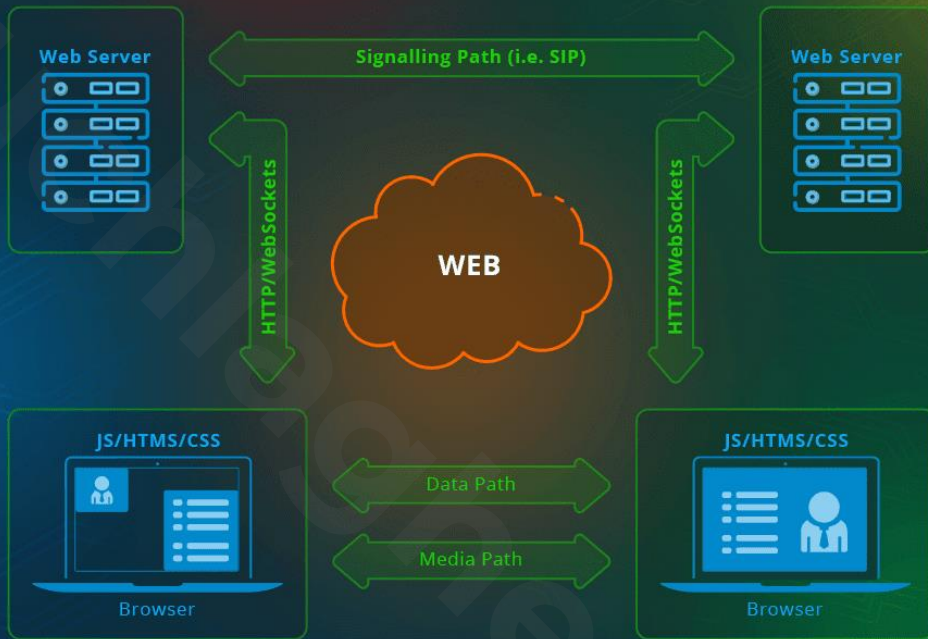
```
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.inviteclientcontext | closing INVITE session m8dl1mthtkin1tel2enrv199k44e | sip:0.15.1.js:18080
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | closing PeerConnection | sip:0.15.1.js:18080
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.invitecontext.sessionDescriptionHandler | m8dl1mthtkin1tel2enrv199k44e | resetIceGatheringComplete | sip:0.15.1.js:18080
Thu Feb 16 2023 21:57:04 GMT+0100 (Central European Standard Time) sip.invite-dialog | INVITE dialog m8dl1mthtkin1tel2enrv199k44e65Q4b/7UNHk3D destroyed | sip:0.15.1.js:18080
```

✓ Here when the call ends we can see that we received a BYE request and the call has ended

## 4 ) Where does the media flow go ?:

- ✓ In this schema we have a resume of the media flow or all the exchanges that start from a client that is using a web browser and interacted with the page here the JavaScript works and send it to the server than the server check for the number call if it available than it start the call to the receiver so its goes from our machine to our router than our router route it to the internet in the direction of the web server than to the colleague

### How does WebRTC work?



mobidev