opensips Summit 2025

WebRTC & The Last Mile

Conrad De Wet | SIPERB

Conrad de Wet, Who am I?

- **2010** – Built a Cloud PBX called Euphoria Telecom.
- Currently has +70K extensions

- **December 2021** – Released **Browser Phone** – Fully featured SIP based WebRTC Phone for Asterisk & FreeSWITCH. https://github.com/Siperb/Browser-Phone

- **March 2024** – Founded **SIPERB** LTD to commercially Provide Browser Phone as a service.
- Available on both **Web** & **Mobile** (OpenSIPS Proxy)
- https://www.siperb.com/

SIPERB

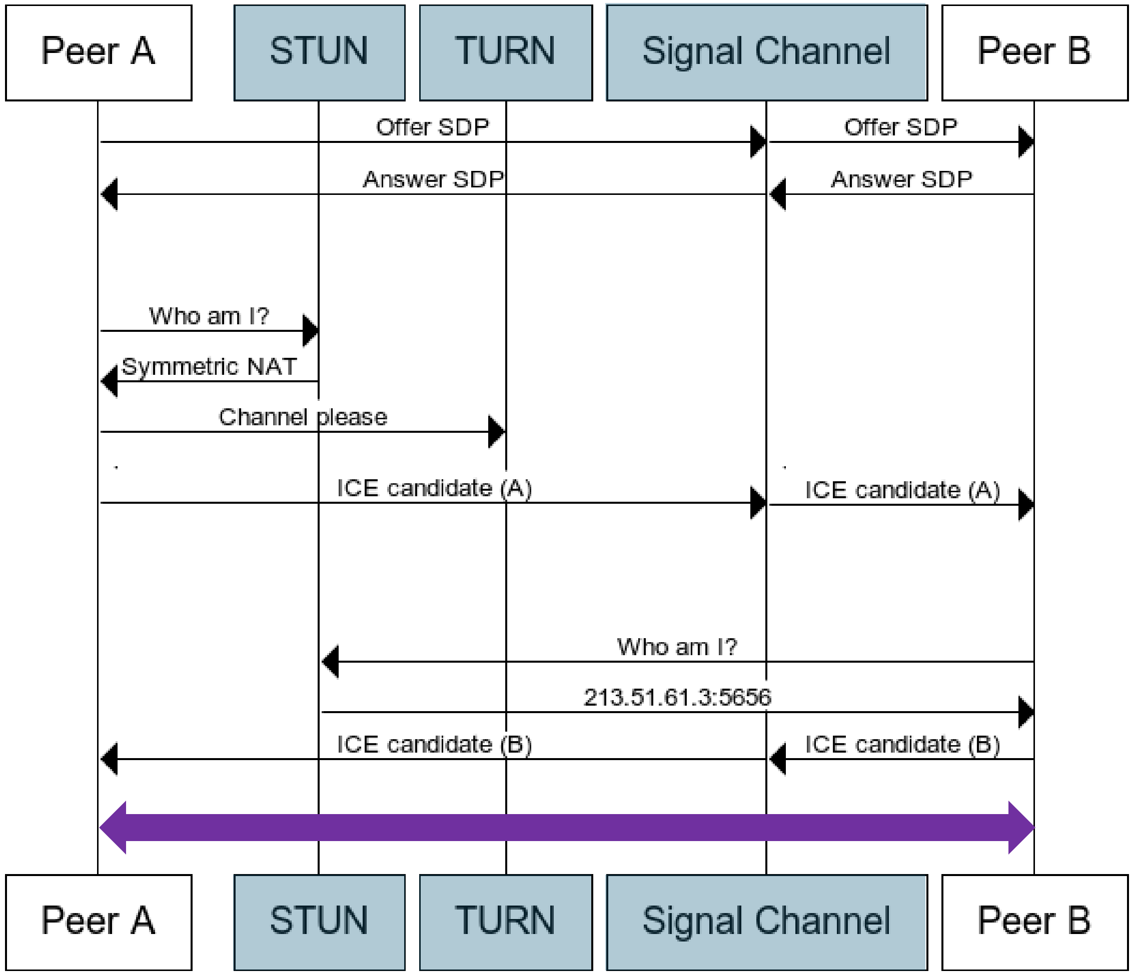
The Problem

- **WebRTC (SIP) ≠ Media (SDP)**
  OpenSIPS sees and handles SIP, but media flows end-to-end.

- **Last Mile Variables**
  ICE, NAT, Jitter, Packet Loss, QOS
  Switch over from Wi-Fi to LAN
  Switch over from Wi-Fi to 4/5G

- **No Built-In Visibility**
  Peer-to-peer media = little or no visibility.

- **Troubleshooting??**
  What tools do we have?
  How do we trouble shoot this?
  What can we do?



Breakdown

- **OpenSIPS - Ping**
  - OPTIONS messages from OpenSIPS to WebRTC client
  - Recording and display this information

- **RTCPeerConnection - GetStats()**
  - Network Address Translation (NAT)
  - Interactive Connectivity Establishment (ICE)
  - Bitrate, Jitter & QOS
  - What control do we have?
  - Recording and display this information

- **Network Failure or Switch-Over Handling**
  - Handing the WebSocket Connection
  - RE-INVITE demo

OpenSIPS - Ping

- OPTIONS messages from OpenSIPS to WebRTC client
  - **nathelper** module

```
modparam("nathelper", "natping_interval", 120)
modparam("nathelper", "ping_nated_only", 0)
modparam("nathelper", "received_avp", "$avp(received)")
modparam("nathelper", "sipping_bflag", "SIP_PING_FLAG")
modparam("nathelper", "sipping_latency_flag", "SIPPING_CALC_LATENCY")
modparam("nathelper", "sipping_from", "sip:hello@ws-eu-west-1.siperb.com")
modparam("nathelper", "sipping_method", "OPTIONS")
modparam("nathelper", "remove_on_timeout_bflag", "SIP_PING_RTN")
modparam("nathelper", "ping_threshold", 3)
modparam("nathelper", "max_pings_lost", 5)
modparam("nathelper", "natping_tcp", 1)
```

OpenSIPS - Ping

- OPTIONS messages from OpenSIPS to WebRTC client
  - route[**register**]

```
# Flag the bflag for OPTIONS PING
setbflag("SIP_PING_FLAG");

# Flag the bflag for removal on lost PINGS
setbflag("SIP_PING_RTN");

# Allow us to measure latency
setbflag("SIPPING_CALC_LATENCY");
```

  - event_route[**E_UL_LATENCY_UPDATE**]

```
event_route[E_UL_LATENCY_UPDATE]{
  $var(contact-instance) = $(param(uri){uri.user});
  xlog("Latency: $param(latency)microseconds”);
  # Save to storage!!
}
```

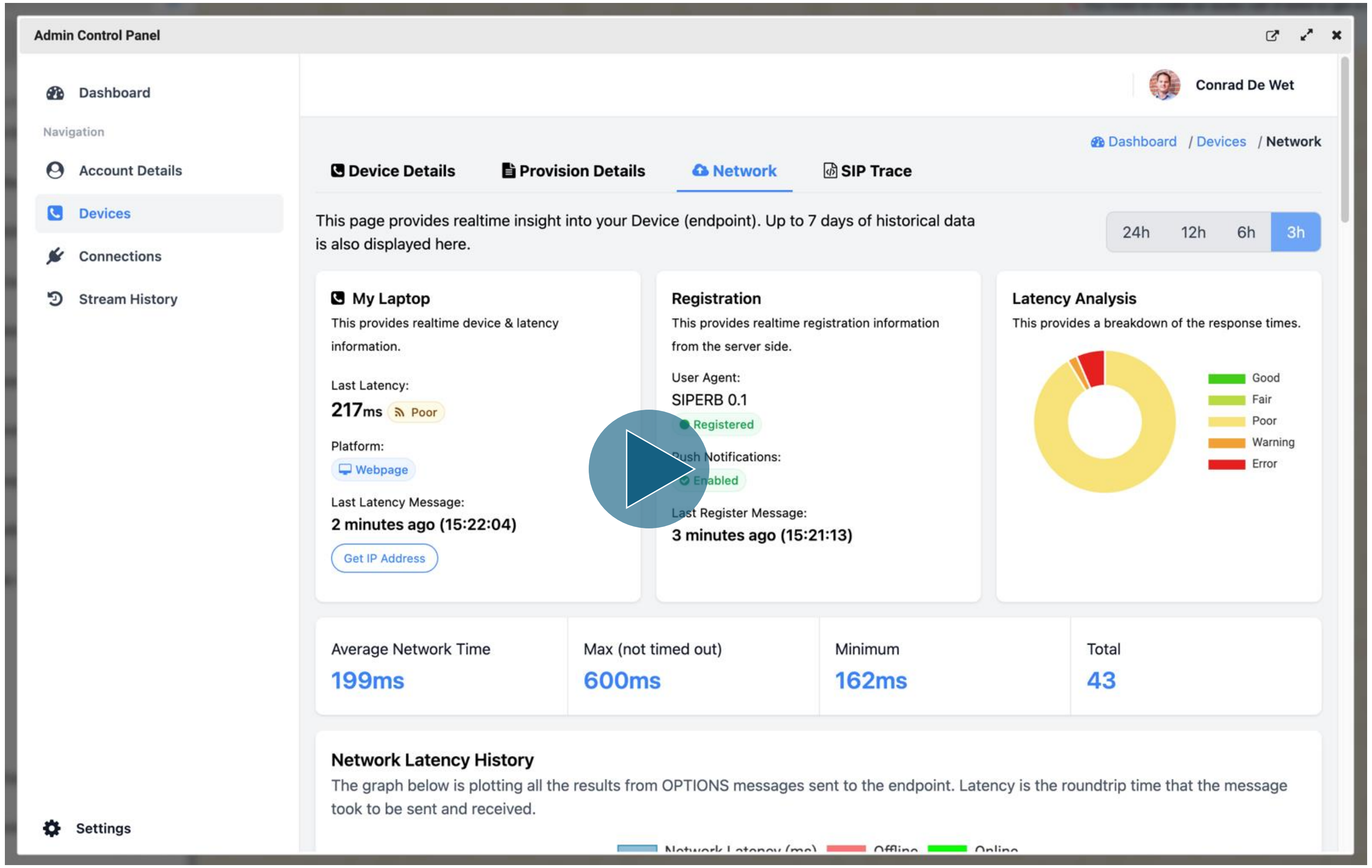
OpenSIPS - Ping

- OPTIONS messages from OpenSIPS to WebRTC client
  - Message Structure

```
OPTIONS sip:alice@192.0.2.143;transport=wss SIP/2.0
Via: SIP/2.0/WS 172.31.10.183:8080;branch=z9hG4bK3416.123
From: <sip:hello@ws-eu-west-1.siperb.com>;tag=e2565de4
To: <sip:alice@192.0.2.143;transport=wss>
Call-ID: 03849a54-18c6aed6-c41dc2
CSeq: 1 OPTIONS
Max-Forwards: 70
Content-Length: 0
```

OpenSIPS - Ping

- Recording and display this information



Admin Control Panel

Conrad De Wet

- Dashboard

Navigation

- Account Details
- Devices
- Connections
- Stream History

Dashboard / Devices / Network

Device Details | **Provision Details** | **Network** | **SIP Trace**

This page provides realtime insight into your Device (endpoint). Up to 7 days of historical data is also displayed here.

24h | 12h | 6h | 3h

My Laptop

This provides realtime device & latency information.

Last Latency:

217ms Poor

Platform:

Webpage

Last Latency Message:

2 minutes ago (15:22:04)

Get IP Address

Registration

This provides realtime registration information from the server side.

User Agent:

SIPERB 0.1

Registered

Push Notifications:

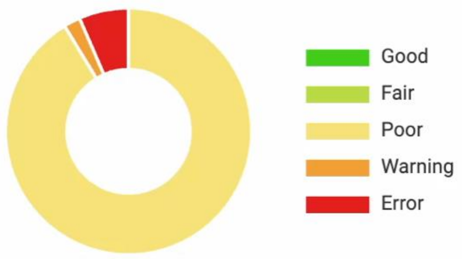
Enabled

Last Register Message:

3 minutes ago (15:21:13)

Latency Analysis

This provides a breakdown of the response times.



Average Network Time	Max (not timed out)	Minimum	Total
199ms	600ms	162ms	43

Network Latency History

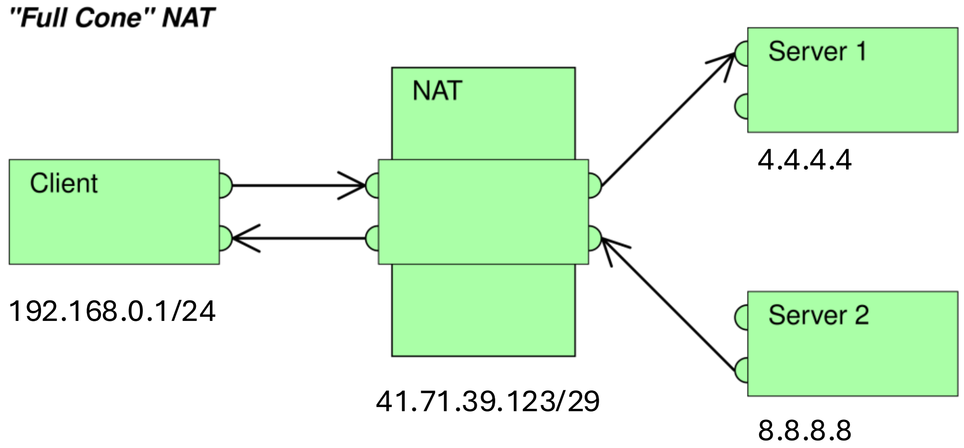
The graph below is plotting all the results from OPTIONS messages sent to the endpoint. Latency is the roundtrip time that the message took to be sent and received.

Network Latency (ms) Offline Online

Settings

RTCPeerConnection - GetStats()

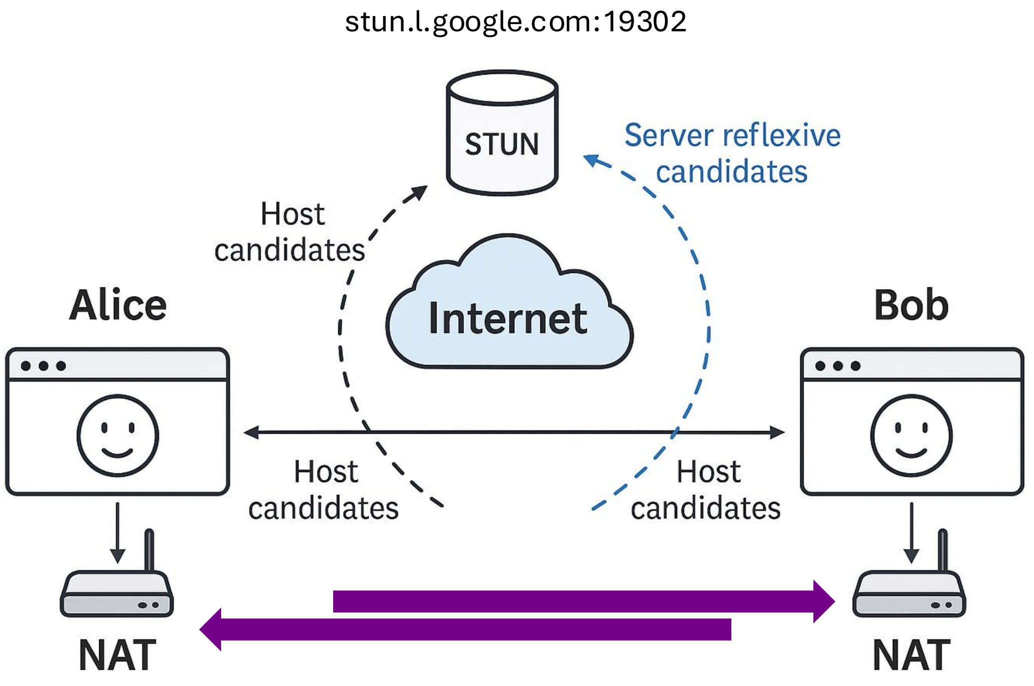
- **Network Address Translation (NAT)**
  - 99.9% chance of NAT with WebRTC (except for LAN deployments)
  - WebRTC SIP Signalling is not subject to NAT problems, due to its stateful TCP connection.
  - WebRTC Media does **not** follow the signalling path, and requires additional information to create a full bidirectional media flow.
  - How do we know our Live IP address?



RTCPeerConnection - GetStats()

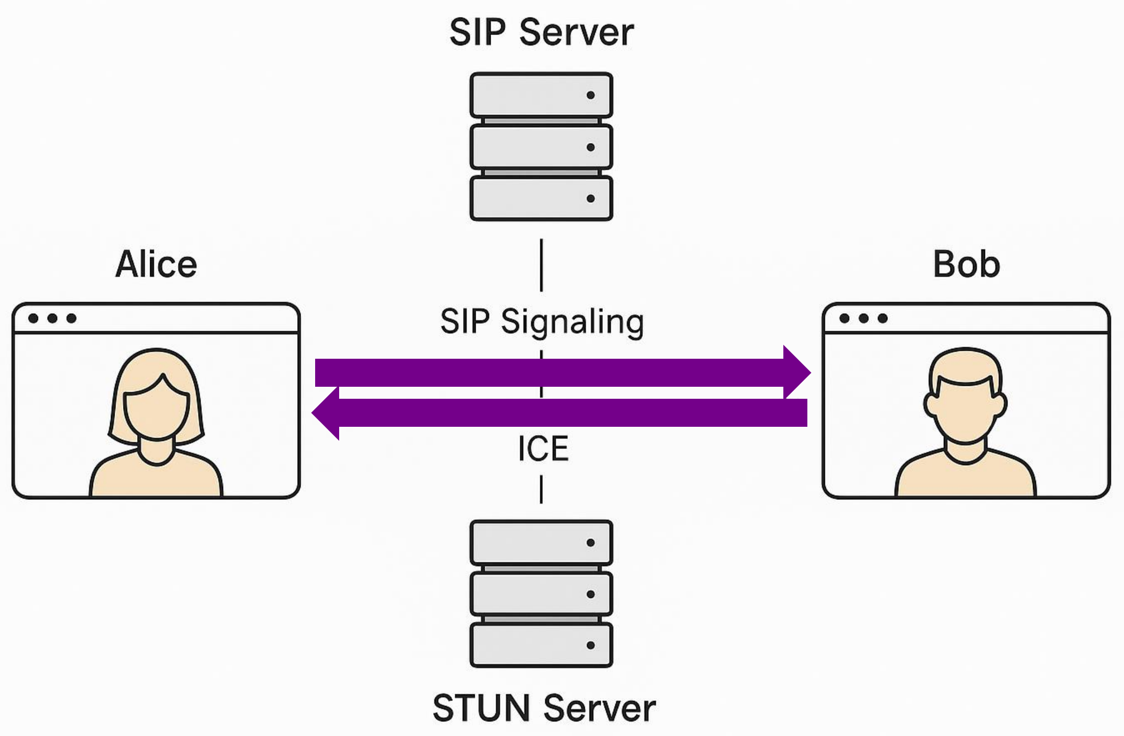
- Interactive Connectivity Establishment (ICE)
  - Used during INVITE only
  - Forms part of the SDP (not SIP headers)
  - Requires:
    - Session Traversal Utilities for NAT (STUN),

    or
    - Traversal Using Relays around NAT (TURN)
      Should be a last resort as media traverses this device, adding latency.



RTCPeerConnection - GetStats()

- Media flows Peer-to-Peer according to what the ICE candidate response is.
  - In WebRTC, we use the **RTCPeerConnection()**
  - Sets the SDP
  - .GetStats()
    - **Bit/Packet Rate**
    - **Jitter**
    - **Round Trip Time**
    - **Packet Loss**
    - **Calculate: QOS/MOS**



RTCPeerConnection - GetStats()

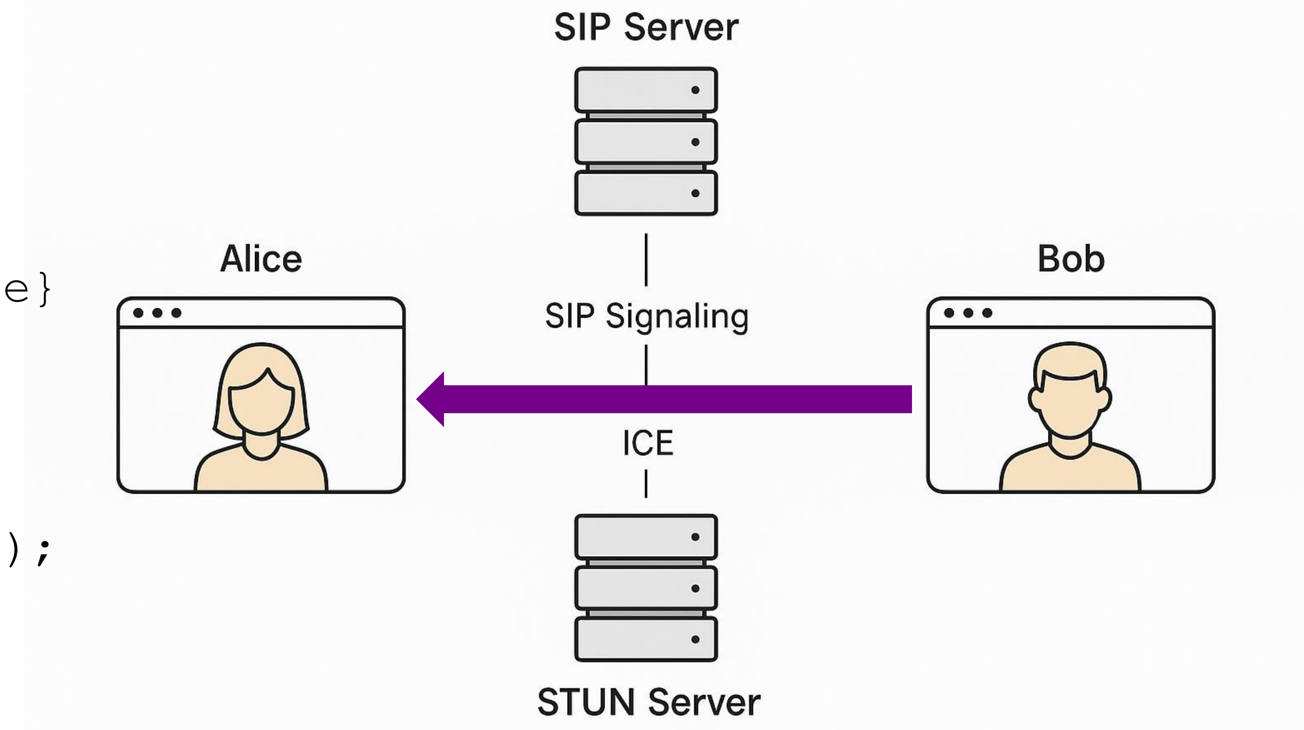
- What control do we have?

- **RTCPeerConnection()**
- .RTCRtpSender()
  - .setParameters({...})

```
{params.encodings[0].maxBitrate = bitrate}
```

Or

```
sender.track.applyConstraints({ height });
```



RTCPeerConnection - GetStats()

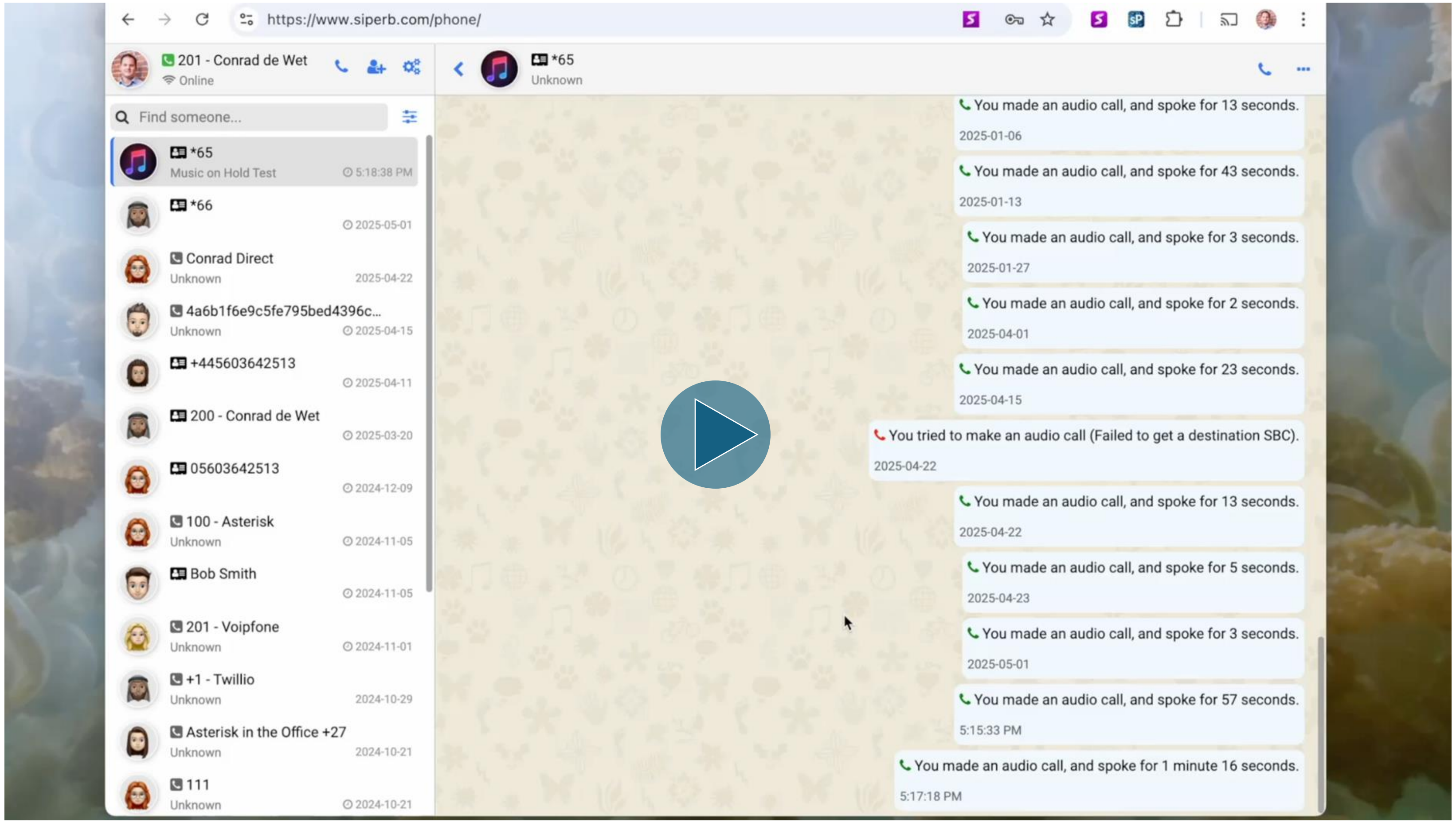
```
const pc = new RTCPeerConnection();
// ... Assume peerconnection is already connected and streaming

setInterval(async () => {
  const stats = await pc.getStats(); // Get the stats from the entire RTCPeerConnection
  stats.forEach(report => {
    // ... Collect the stats you want.
    console.log(`Packets Received: ${report.packetsReceived}`);
    console.log(`Round Trip Time: ${report.roundTripTime}`);
    console.log(`Packets Lost: ${report.packetsLost}`);
    console.log(`Jitter: ${report.jitter}`);

  });
}, 1000);
```

RTCPeerConnection - GetStats()

- Recording and display this information



https://www.siperb.com/phone/

201 - Conrad de Wet
Online

Find someone...

*65	Music on Hold Test	5:18:38 PM
*66		2025-05-01
Conrad Direct	Unknown	2025-04-22
4a6b1f6e9c5fe795bed4396c...	Unknown	2025-04-15
+445603642513		2025-04-11
200 - Conrad de Wet		2025-03-20
05603642513		2024-12-09
100 - Asterisk	Unknown	2024-11-05
Bob Smith		2024-11-05
201 - Voipfone	Unknown	2024-11-01
+1 - Twillio	Unknown	2024-10-29
Asterisk in the Office +27	Unknown	2024-10-21
111	Unknown	2024-10-21

*65
Unknown

You made an audio call, and spoke for 13 seconds.
2025-01-06

You made an audio call, and spoke for 43 seconds.
2025-01-13

You made an audio call, and spoke for 3 seconds.
2025-01-27

You made an audio call, and spoke for 2 seconds.
2025-04-01

You made an audio call, and spoke for 23 seconds.
2025-04-15

You tried to make an audio call (Failed to get a destination SBC).
2025-04-22

You made an audio call, and spoke for 13 seconds.
2025-04-22

You made an audio call, and spoke for 5 seconds.
2025-04-23

You made an audio call, and spoke for 3 seconds.
2025-05-01

You made an audio call, and spoke for 57 seconds.
5:15:33 PM

You made an audio call, and spoke for 1 minute 16 seconds.
5:17:18 PM

Network Failure or Switch-Over Handling

- Handing the WebSocket Connection
  - SIP.js – Version >= 0.20.0

```
let ws = new WebSocket("wss://ws-eu-west-1.siperb.com:4443/ws", "sip");
ws.addEventListener("close", (ev) => this.onWebSocketClose(ev, ws));
ws.addEventListener("error", (ev) => this.onWebSocketError(ev, ws));
ws.addEventListener("open", (ev) => this.onWebSocketOpen(ev, ws));
ws.addEventListener("message", (ev) => this.onWebSocketMessage(ev, ws));
```

  - Error event only fires on server error, and can take up to 30 seconds to fire when transitioning networks.

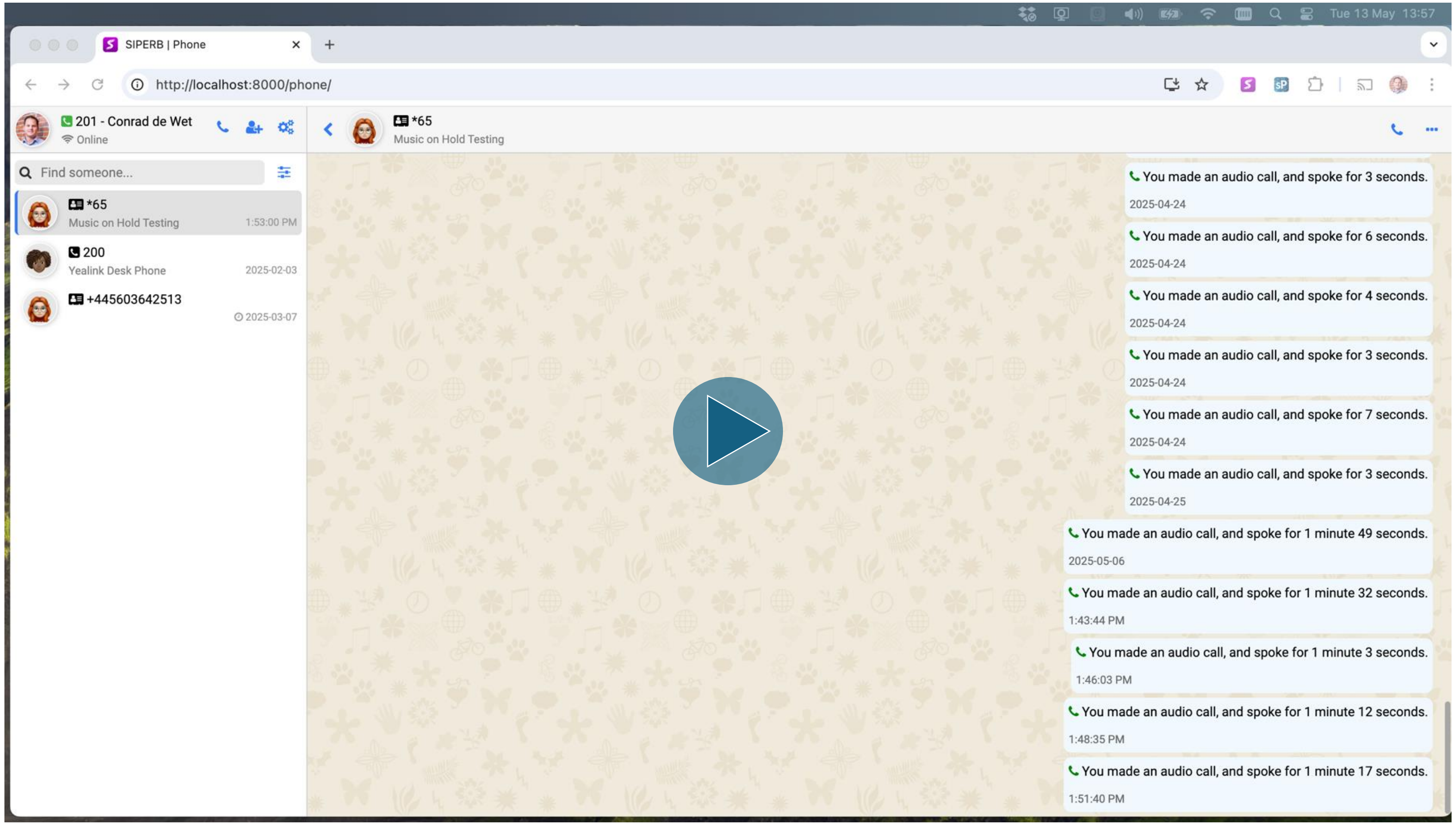
```
window.addEventListener("offline", (event) => {

});
window.addEventListener("online", (event) => {

});
```

Network Failure or Switch-Over Handling

- RE-INVITE demo



Tue 13 May 13:51

SIPERB | Phone

http://localhost:8000/phone/

201 - Conrad de Wet
Online

Find someone...

*65
Music on Hold Testing 1:49:50 PM

200
Yealink Desk Phone 2025-02-03

+445603642513
2025-03-07

*65
Music on Hold Testing

You made an audio call, and spoke for 2 seconds.
2025-04-24

You made an audio call, and spoke for 3 seconds.
2025-04-24

You made an audio call, and spoke for 6 seconds.
2025-04-24

You made an audio call, and spoke for 4 seconds.
2025-04-24

You made an audio call, and spoke for 3 seconds.
2025-04-24

You made an audio call, and spoke for 7 seconds.
2025-04-24

You made an audio call, and spoke for 3 seconds.
2025-04-25

You made an audio call, and spoke for 1 minute 49 seconds.
2025-05-06

You made an audio call, and spoke for 1 minute 32 seconds.
1:43:44 PM

You made an audio call, and spoke for 1 minute 3 seconds.
1:46:03 PM

You made an audio call, and spoke for 1 minute 12 seconds.
1:48:35 PM

Thank you!

- **Questions?**

- **Information & Downloads:**

  - **Siperb Web Site**: https://www.siperb.com/
  - **Softphone Download**: https://www.siperb.com/phone/
  - **LinkedIn** : https://linkedin.com/company/siperb/
  - **Twitter (X)** : https://x.com/SiperbDotCom/
  - **Facebook** : https://www.facebook.com/siperbltd/
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