Bridging WebRTC and SIP using GStreamer & SIPjs

Sanchayan Maity

Who?

- Open source consulting firm based out of Bangalore and Toronto.

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- Building high-quality, low-level systems software.

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- Building high-quality, low-level systems software.
- Most of our work revolves around audio/video using GStreamer and PipeWire.

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- Carried by transport layer protocols like TCP or UDP

Why SIP?

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- Dial in and dial out

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- SIP is how phones connect to the Internet
- Dial in and dial out
- Conference room equipment interoperates using SIP

SIP session setup

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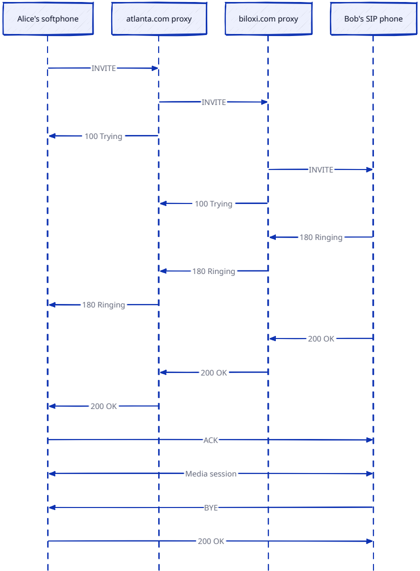

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  - Uses WebSockets for SIP signalling (RFC 7118)
  - Uses `RTCPeerConnection` for media
  - Primarily used on browser

SIPjs - User agent/Registration[1]

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```
const transportOptions = {
  server: "wss://atlanta.com:8443"
};
const uri = UserAgent.makeURI("sip:alice@atlanta.com");
const userAgentOptions: UserAgentOptions = {
  authorizationPassword: 'secretPassword',
  authorizationUsername: 'authorizationUsername',
  transportOptions,
  uri
};
const userAgent = new UserAgent(userAgentOptions);
userAgent.start().then(() => {
  registerer.register();
});
```

[1] SIPjs guides - user agent

SIPjs - make or receive a call[2][3]

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```
// userAgent defined elsewhere
userAgent.start().then(() => {
  const target = UserAgent.makeURI("sip:bob@biloxi.com");

  const inviter = new Inviter(userAgent, target);
  // make a call
  inviter.invite();
});

// receive call
function onInvite(invitation) {
  invitation.accept();
}
```

[2] SIPjs guides - make call
[3] SIPjs guides - receive call

SIPjs - attach media[4]

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```
// Assumes you have a media element on the DOM
const mediaElement = document.getElementById('mediaElement');

const remoteStream = new MediaStream(); // getUserMedia
function setupRemoteMedia(session: Session) {
  session.sessionDescriptionHandler.peerConnection.getReceivers().
    forEach((receiver) => {
      if (receiver.track) {
        remoteStream.addTrack(receiver.track);
      }
  });
  mediaElement.srcObject = remoteStream;
  mediaElement.play();
}
```

[4] SIPjs guides - attach media

SIPjs - Session Description Handler Interface

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```
export interface SessionDescriptionHandler {
  getDescription(
    options?: SessionDescriptionHandlerOptions,
    modifiers?: Array<SessionDescriptionHandlerModifier>
  ): Promise<BodyAndContentType>;

  hasDescription(contentType: string): boolean;

  setDescription(
    sdp: string,
    options?: SessionDescriptionHandlerOptions,
    modifiers?: Array<SessionDescriptionHandlerModifier>
  ): Promise<void>;
}
```

SIPjs - Session Description Handler

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```
export class SessionDescriptionHandler
  implements SessionDescriptionHandlerDefinition {
    protected _localMediaStream: MediaStream;
    protected _remoteMediaStream: MediaStream;
    protected _dataChannel: RTCDataChannel;
    protected _peerConnection: RTCPeerConnection;

    private iceGatheringCompletePromise: Promise<void>;
    private iceGatheringCompleteTimeoutId: number;
    private iceGatheringCompleteResolve: ResolveFunction;
    private iceGatheringCompleteReject: RejectFunction;
    private localMediaStreamConstraints: MediaStreamConstraints;
    private onDataChannel: ((dataChannel: RTCDataChannel) => void);
}
```

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- `SIPjs` provides hooks to override `RTCPeerConnection` usage
  - GStreamer has a `webrtc` implementation with an API similar to `RTCPeerConnection`
  - Use this to call out to a GStreamer-based implementation leveraging `webrtcbin`
  - that's `SIPjs` (Node) -> `GStreamer` app (Rust)

webrtcbin <-> sipjs - I

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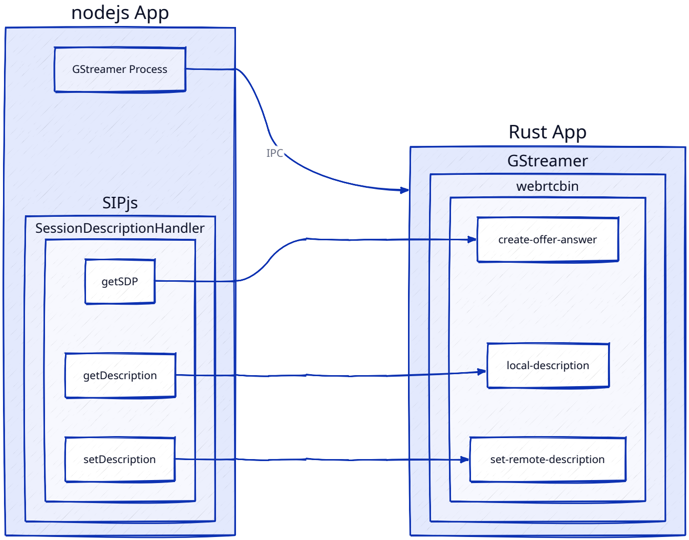

webrtcbin <-> SIPjs - II

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```
class GstSessionDescriptionHandler implements SessionDescriptionHandler {
  gstProcess: ChildProcess;

  private async sendGst(msg: JsMessage): Promise<GstMessageSdp> {
    this.gstProcess.send(msg);
    return new Promise((resolve, reject) => {
      this.gstProcess.stdout.on('data', (message: string) => {
        let msg: GstMessage = JSON.parse(message);
        resolve(msg);
      });
    });
  }

  hasDescription(contentType: string): boolean {
    return contentType === 'application/sdp';
  }
```

webrtcbin <-> SIPjs - III

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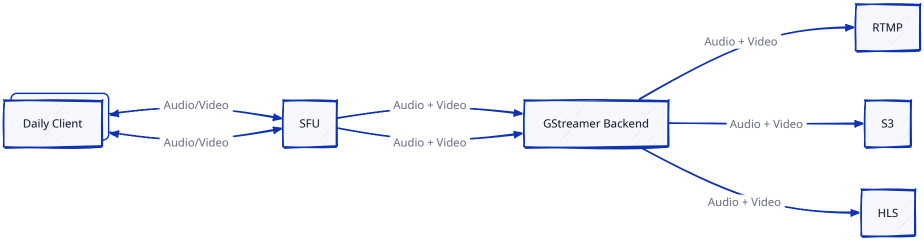
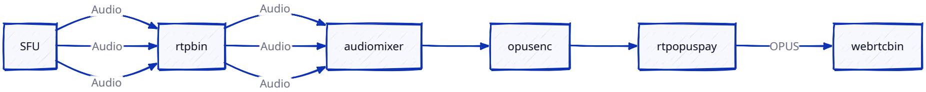
```
async getDescription(): Promise<BodyAndContentType> {
  let msg =
    await this.sendGst({ code: JsMessageCode.GET_LOCAL_DESCRIPTION });
  return Promise.resolve({
    body: msg.sdpBody,
    contentType: 'application/sdp',
  });
}

async setDescription(
  sdp: string,
): Promise<void> {
  await this.sendGst({ code: JsMessageCode.SET_REMOTE_DESCRIPTION,
  sdpBody: sdp });
}
```

Daily back-end architecture[5]



[5] GStreamer for your back-end services

WebRTC -> SIP



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Audio
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OPUS
webrtcbin

SIP -> WebRTC

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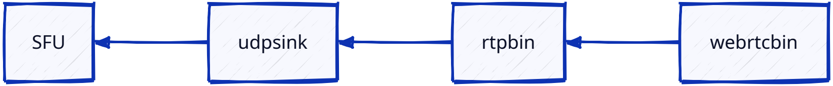

SIP Architecture

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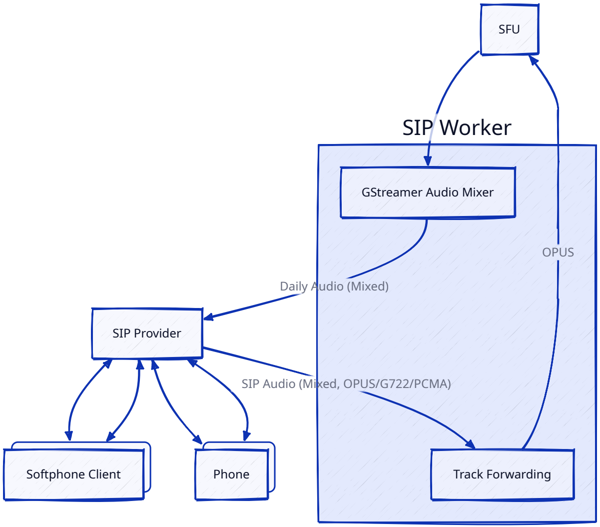

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- Implement matrix/audio mixing ourselves
- Add video support

Questions?

Thank You.