Custom Audio Streaming app (SSE)

This article overviews the server and client code for a custom asynchronous web app built with ADK Streaming and FastAPI, enabling real-time, bidirectional audio and text communication with Server-Sent Events (SSE). The key features are:

Server-Side (Python/FastAPI): - FastAPI + ADK integration - Server-Sent Events for real-time streaming - Session management with isolated user contexts - Support for both text and audio communication modes - Google Search tool integration for grounded responses

Client-Side (JavaScript/Web Audio API): - Real-time bidirectional communication via SSE and HTTP POST - Professional audio processing using AudioWorklet processors - Seamless mode switching between text and audio - Automatic reconnection and error handling - Base64 encoding for audio data transmission

1. Install ADK

Create & Activate Virtual Environment (Recommended):

```
# Create
python -m venv .venv
# Activate (each new terminal)
# macOS/Linux: source .venv/bin/activate
# Windows CMD: .venv\Scripts\activate.bat
# Windows PowerShell: .venv\Scripts\Activate.ps1
```

Install ADK:

```
pip install google-adk==1.0.0
```

Set `SSL_CERT_FILE` variable with the following command.

```
export SSL_CERT_FILE=$(python -m certifi)
```

Download the sample code:

```
git clone --no-checkout https://github.com/google/adk-docs.git
cd adk-docs
git sparse-checkout init --cone
git sparse-checkout set examples/python/snippets/streaming/adk-streaming
git checkout main
cd examples/python/snippets/streaming/adk-streaming/app
```

This sample code has the following files and folders:

```
adk-streaming/
└── app/ # the web app folder
    ├── .env # Gemini API key / Google Cloud Project ID
    ├── main.py # FastAPI web app
    ├── static/ # Static content folder
    |   ├── js # JavaScript files folder (includes app.js)
    |   └── index.html # The web client page
    └── google_search_agent/ # Agent folder
        ├── __init__.py # Python package
        └── agent.py # Agent definition
```

2. Set up the platform

To run the sample app, choose a platform from either Google AI Studio or Google Cloud Vertex AI:

Gemini - Google AI Studio

1. Get an API key from Google AI Studio.
2. Open the `.env` file located inside (`app/`) and copy-paste the following code.

.env

```
GOOGLE_GENAI_USE_VERTEXAI=FALSE
GOOGLE_API_KEY=PASTE_YOUR_ACTUAL_API_KEY_HERE
```

3. Replace `PASTE_YOUR_ACTUAL_API_KEY_HERE` with your actual `API KEY`.

Gemini - Google Cloud Vertex AI

1. You need an existing Google Cloud account and a project.

- Set up a Google Cloud project
- Set up the gcloud CLI
- Authenticate to Google Cloud, from the terminal by running `gcloud auth login`.
- Enable the Vertex AI API.

2. Open the `.env` file located inside (`app/`). Copy-paste the following code and update the project ID and location.

.env

```
GOOGLE_GENAI_USE_VERTEXAI=TRUE
GOOGLE_CLOUD_PROJECT=PASTE_YOUR_ACTUAL_PROJECT_ID
GOOGLE_CLOUD_LOCATION=us-central1
```

3. Interact with Your Streaming app

1. **Navigate to the Correct Directory:**

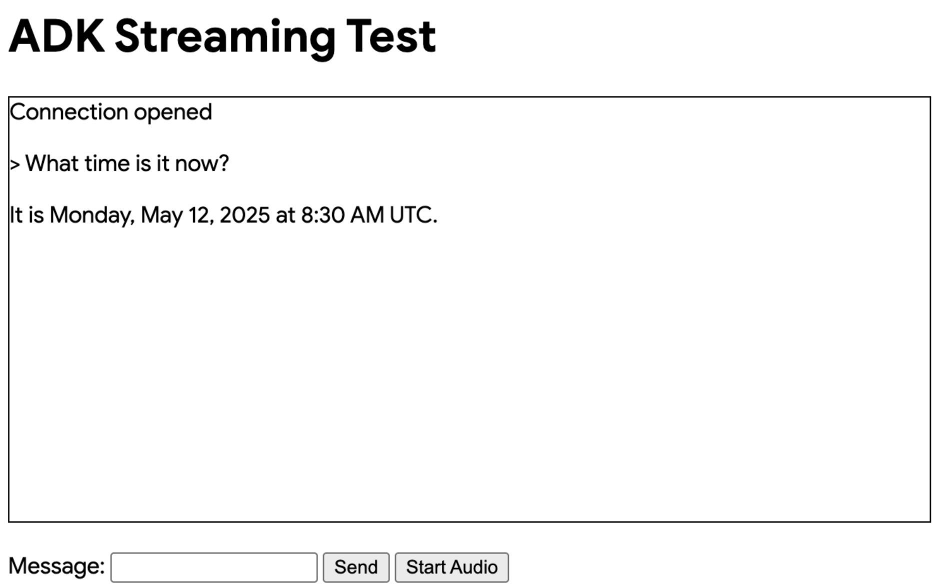
To run your agent effectively, make sure you are in the **app folder (`adk-streaming/app`)**

2. **Start the Fast API**: Run the following command to start CLI interface with

```
uvicorn main:app --reload
```

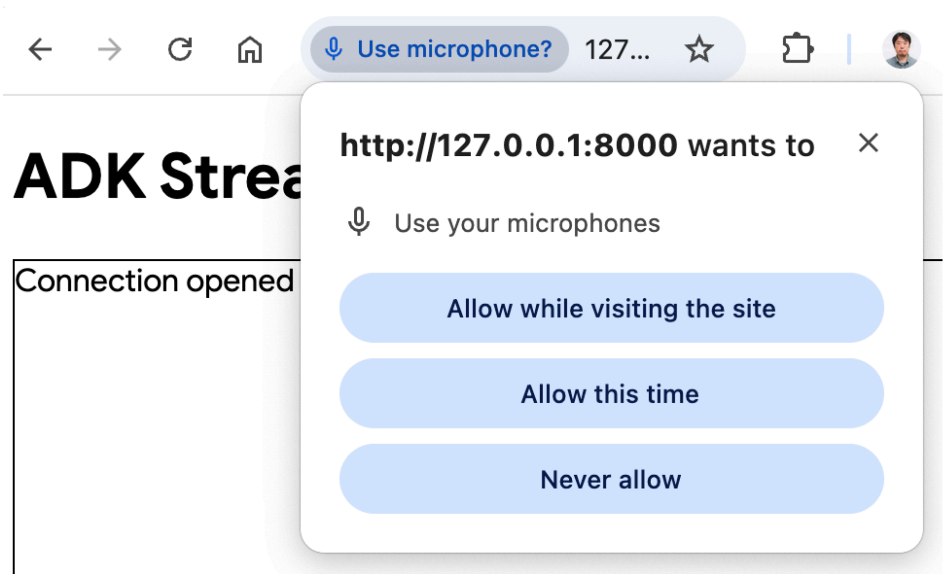
3. **Access the app with the text mode:** Once the app starts, the terminal will display a local URL (e.g., http://localhost:8000). Click this link to open the UI in your browser.

Now you should see the UI like this:

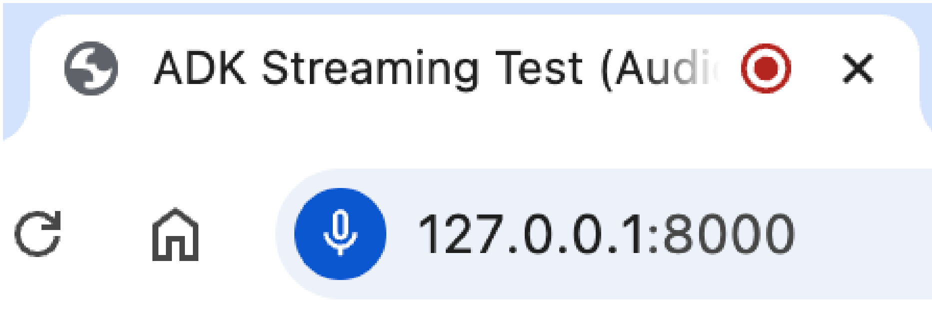


Try asking a question `What time is it now?`. The agent will use Google Search to respond to your queries. You would notice that the UI shows the agent's response as streaming text. You can also send messages to the agent at any time, even while the agent is still responding. This demonstrates the bidirectional communication capability of ADK Streaming.

4. **Access the app with the audio mode:** Now click the `Start Audio` button. The app reconnects with the server in an audio mode, and the UI will show the following dialog for the first time:



Click `Allow while visiting the site`, then you will see the microphone icon will be shown at the top of the browser:



Now you can talk to the agent with voice. Ask questions like `What time is it now?` with voice and you will hear the agent responding in voice too. As Streaming for ADK supports multiple languages, it can also respond to question in the supported languages.

5. **Check console logs**

If you are using the Chrome browser, use the right click and select `Inspect` to open the DevTools. On the `Console`, you can see the incoming and outgoing audio data such as `[CLIENT TO AGENT]` and `[AGENT TO CLIENT]`, representing the audio data streaming in and out between the browser and the server.

At the same time, in the app server console, you should see something like this:

```
Client #90766266 connected via SSE, audio mode: false
INFO:     127.0.0.1:52692 - "GET /events/90766266?
is_audio=false HTTP/1.1" 200 OK
[CLIENT TO AGENT]: hi
INFO:     127.0.0.1:52696 - "POST /send/90766266 HTTP/1.1" 200
OK
[AGENT TO CLIENT]: text/plain: {'mime_type': 'text/plain',
'data': 'Hi'}
[AGENT TO CLIENT]: text/plain: {'mime_type': 'text/plain',
'data': ' there! How can I help you today?\n'}
[AGENT TO CLIENT]: {'turn_complete': True, 'interrupted': None}
```

These console logs are important in case you develop your own streaming application. In many cases, the communication failure between the browser and server becomes a major cause for the streaming application bugs.

6. **Troubleshooting tips**

- **When `gemini-2.0-flash-exp` model doesn't work:** If you see any errors on the app server console with regard to `gemini-2.0-flash-exp` model availability, try replacing it with `gemini-2.0-flash-live-001` on `app/google_search_agent/agent.py` at line 6.

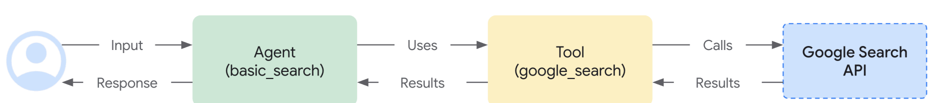
4. Agent definition

The agent definition code `agent.py` in the `google_search_agent` folder is where the agent's logic is written:

```
from google.adk.agents import Agent
from google.adk.tools import google_search  # Import the tool

root_agent = Agent(
   name="google_search_agent",
   model="gemini-2.0-flash-exp", # if this model does not work,
try below
   #model="gemini-2.0-flash-live-001",
   description="Agent to answer questions using Google
Search.",
   instruction="Answer the question using the Google Search
tool.",
   tools=[google_search],
)
```

Notice how easily you integrated grounding with Google Search capabilities. The `Agent` class and the `google_search` tool handle the complex interactions with the LLM and grounding with the search API, allowing you to focus on the agent's *purpose* and *behavior*.



The server and client architecture enables real-time, bidirectional communication between web clients and AI agents with proper session isolation and resource management.

5. Server side code overview

The FastAPI server provides real-time communication between web clients and the AI agent.

Bidirectional communication overview

Client-to-Agent Flow:

1. **Connection Establishment** - Client opens SSE connection to `/events/{user_id}`, triggering session creation and storing request queue in `active_sessions`
2. **Message Transmission** - Client sends POST to `/send/{user_id}` with JSON payload containing `mime_type` and `data`
3. **Queue Processing** - Server retrieves session's `live_request_queue` and forwards message to agent via `send_content()` or `send_realtime()`

Agent-to-Client Flow:

1. **Event Generation** - Agent processes requests and generates events through `live_events` async generator
2. **Stream Processing** - `agent_to_client_sse()` filters events and formats them as SSE-compatible JSON
3. **Real-time Delivery** - Events stream to client via persistent HTTP connection with proper SSE headers

Session Management:

- **Per-User Isolation** - Each user gets unique session stored in `active_sessions` dict
- **Lifecycle Management** - Sessions auto-cleanup on disconnect with proper resource disposal
- **Concurrent Support** - Multiple users can have simultaneous active sessions

Error Handling:

- **Session Validation** - POST requests validate session existence before processing
- **Stream Resilience** - SSE streams handle exceptions and perform cleanup automatically

- **Connection Recovery** - Clients can reconnect by re-establishing SSE connection

Agent Session Management

The `start_agent_session()` function creates isolated AI agent sessions:

```
async def start_agent_session(user_id, is_audio=False):
    """Starts an agent session"""

    # Create a Runner
    runner = InMemoryRunner(
        app_name=APP_NAME,
        agent=root_agent,
    )

    # Create a Session
    session = await runner.session_service.create_session(
        app_name=APP_NAME,
        user_id=user_id,  # Replace with actual user ID
    )

    # Set response modality
    modality = "AUDIO" if is_audio else "TEXT"
    run_config = RunConfig(response_modalities=[modality])

    # Create a LiveRequestQueue for this session
    live_request_queue = LiveRequestQueue()

    # Start agent session
    live_events = runner.run_live(
        session=session,
        live_request_queue=live_request_queue,
        run_config=run_config,
    )
    return live_events, live_request_queue
```

- **InMemoryRunner Setup** - Creates a runner instance that manages the agent lifecycle in memory, with the app name "ADK Streaming example" and the Google Search agent.
- **Session Creation** - Uses `runner.session_service.create_session()` to establish a unique session per user ID, enabling multiple concurrent users.
- **Response Modality Configuration** - Sets `RunConfig` with either "AUDIO" or "TEXT" modality based on the `is_audio` parameter, determining output format.

- **LiveRequestQueue** - Creates a bidirectional communication channel that queues incoming requests and enables real-time message passing between client and agent.
- **Live Events Stream** - `runner.run_live()` returns an async generator that yields real-time events from the agent, including partial responses, turn completions, and interruptions.

Server-Sent Events (SSE) Streaming

The `agent_to_client_sse()` function handles real-time streaming from agent to client:

```
async def agent_to_client_sse(live_events):
    """Agent to client communication via SSE"""
    async for event in live_events:
        # If the turn complete or interrupted, send it
        if event.turn_complete or event.interrupted:
            message = {
                "turn_complete": event.turn_complete,
                "interrupted": event.interrupted,
            }
            yield f"data: {json.dumps(message)}\n\n"
            print(f"[AGENT TO CLIENT]: {message}")
            continue

        # Read the Content and its first Part
        part: Part = (
            event.content and event.content.parts and
event.content.parts[0]
        )
        if not part:
            continue

        # If it's audio, send Base64 encoded audio data
        is_audio = part.inline_data and
part.inline_data.mime_type.startswith("audio/pcm")
        if is_audio:
            audio_data = part.inline_data and
part.inline_data.data
            if audio_data:
                message = {
                    "mime_type": "audio/pcm",
                    "data":
base64.b64encode(audio_data).decode("ascii")
                }
                yield f"data: {json.dumps(message)}\n\n"
                print(f"[AGENT TO CLIENT]: audio/pcm:
{len(audio_data)} bytes.")
```

```
                continue

        # If it's text and a parial text, send it
        if part.text and event.partial:
            message = {
                "mime_type": "text/plain",
                "data": part.text
            }
            yield f"data: {json.dumps(message)}\n\n"
            print(f"[AGENT TO CLIENT]: text/plain: {message}")
```

- **Event Processing Loop** - Iterates through `live_events` async generator, processing each event as it arrives from the agent.
- **Turn Management** - Detects conversation turn completion or interruption events and sends JSON messages with `turn_complete` and `interrupted` flags to signal conversation state changes.
- **Content Part Extraction** - Extracts the first `Part` from event content, which contains either text or audio data.
- **Audio Streaming** - Handles PCM audio data by:
- Detecting `audio/pcm` MIME type in `inline_data`
- Base64 encoding raw audio bytes for JSON transmission
- Sending with `mime_type` and `data` fields
- **Text Streaming** - Processes partial text responses by sending incremental text updates as they're generated, enabling real-time typing effects.
- **SSE Format** - All data is formatted as `data: {json}\n\n` following SSE specification for browser EventSource API compatibility.

HTTP Endpoints and Routing

Root Endpoint

GET / - Serves `static/index.html` as the main application interface using FastAPI's `FileResponse`.

SSE Events Endpoint

```
@app.get("/events/{user_id}")
async def sse_endpoint(user_id: int, is_audio: str = "false"):
    """SSE endpoint for agent to client communication"""
```

```
    # Start agent session
    user_id_str = str(user_id)
    live_events, live_request_queue = await
start_agent_session(user_id_str, is_audio == "true")

    # Store the request queue for this user
    active_sessions[user_id_str] = live_request_queue

    print(f"Client #{user_id} connected via SSE, audio mode:
{is_audio}")

    def cleanup():
        live_request_queue.close()
        if user_id_str in active_sessions:
            del active_sessions[user_id_str]
        print(f"Client #{user_id} disconnected from SSE")

    async def event_generator():
        try:
            async for data in agent_to_client_sse(live_events):
                yield data
        except Exception as e:
            print(f"Error in SSE stream: {e}")
        finally:
            cleanup()

    return StreamingResponse(
        event_generator(),
        media_type="text/event-stream",
        headers={
            "Cache-Control": "no-cache",
            "Connection": "keep-alive",
            "Access-Control-Allow-Origin": "*",
            "Access-Control-Allow-Headers": "Cache-Control"
        }
    )
```

GET /events/{user_id} - Establishes persistent SSE connection:

- **Parameters** - Takes `user_id` (int) and optional `is_audio` query parameter (defaults to "false")
- **Session Initialization** - Calls `start_agent_session()` and stores the `live_request_queue` in `active_sessions` dict using `user_id` as key
- **StreamingResponse** - Returns `StreamingResponse` with:
- `event_generator()` async function that wraps `agent_to_client_sse()`
- MIME type: `text/event-stream`

- CORS headers for cross-origin access
- Cache-control headers to prevent caching
- **Cleanup Logic** - Handles connection termination by closing the request queue and removing from active sessions, with error handling for stream interruptions.

Message Sending Endpoint

```
@app.post("/send/{user_id}")
async def send_message_endpoint(user_id: int, request:
Request):
    """HTTP endpoint for client to agent communication"""

    user_id_str = str(user_id)

    # Get the live request queue for this user
    live_request_queue = active_sessions.get(user_id_str)
    if not live_request_queue:
        return {"error": "Session not found"}

    # Parse the message
    message = await request.json()
    mime_type = message["mime_type"]
    data = message["data"]

    # Send the message to the agent
    if mime_type == "text/plain":
        content = Content(role="user", parts=
[Part.from_text(text=data)])
        live_request_queue.send_content(content=content)
        print(f"[CLIENT TO AGENT]: {data}")
    elif mime_type == "audio/pcm":
        decoded_data = base64.b64decode(data)

live_request_queue.send_realtime(Blob(data=decoded_data,
mime_type=mime_type))
        print(f"[CLIENT TO AGENT]: audio/pcm:
{len(decoded_data)} bytes")
    else:
        return {"error": f"Mime type not supported:
{mime_type}"}

    return {"status": "sent"}
```

POST /send/{user_id} - Receives client messages:

- **Session Lookup** - Retrieves `live_request_queue` from `active_sessions` or returns error if session doesn't exist

- **Message Processing** - Parses JSON with `mime_type` and `data` fields:
- **Text Messages** - Creates `Content` with `Part.from_text()` and sends via `send_content()`
- **Audio Messages** - Base64 decodes PCM data and sends via `send_realtime()` with `Blob`
- **Error Handling** - Returns appropriate error responses for unsupported MIME types or missing sessions.

6. Client side code overview

The client-side consists of a web interface with real-time communication and audio capabilities:

HTML Interface (`static/index.html`)

```
<!doctype html>
<html>
  <head>
    <title>ADK Streaming Test (Audio)</title>
    <script src="/static/js/app.js" type="module"></script>
  </head>

  <body>
    <h1>ADK Streaming Test</h1>
    <div
      id="messages"
      style="height: 300px; overflow-y: auto; border: 1px solid
black"></div>
    <br />

    <form id="messageForm">
      <label for="message">Message:</label>
      <input type="text" id="message" name="message" />
      <button type="submit" id="sendButton"
disabled>Send</button>
      <button type="button" id="startAudioButton">Start
Audio</button>
    </form>
  </body>

</html>
```

Simple web interface with: - **Messages Display** - Scrollable div for conversation history - **Text Input Form** - Input field and send button for text messages - **Audio Control** - Button to enable audio mode and microphone access

Main Application Logic (`static/js/app.js`)

Session Management (`app.js`)

```
const sessionId = Math.random().toString().substring(10);
const sse_url =
  "http://" + window.location.host + "/events/" + sessionId;
const send_url =
  "http://" + window.location.host + "/send/" + sessionId;
let is_audio = false;
```

- **Random Session ID** - Generates unique session ID for each browser instance
- **URL Construction** - Builds SSE and send endpoints with session ID
- **Audio Mode Flag** - Tracks whether audio mode is enabled

Server-Sent Events Connection (`app.js`)

connectSSE() function handles real-time server communication:

```
// SSE handlers
function connectSSE() {
  // Connect to SSE endpoint
  eventSource = new EventSource(sse_url + "?is_audio=" +
is_audio);

  // Handle connection open
  eventSource.onopen = function () {
    // Connection opened messages
    console.log("SSE connection opened.");
    document.getElementById("messages").textContent =
"Connection opened";

    // Enable the Send button
    document.getElementById("sendButton").disabled = false;
    addSubmitHandler();
  };

  // Handle incoming messages
  eventSource.onmessage = function (event) {
    ...
  };

  // Handle connection close
```

```
  eventSource.onerror = function (event) {
    console.log("SSE connection error or closed.");
    document.getElementById("sendButton").disabled = true;
    document.getElementById("messages").textContent =
"Connection closed";
    eventSource.close();
    setTimeout(function () {
      console.log("Reconnecting...");
      connectSSE();
    }, 5000);
  };
}
```

- **EventSource Setup** - Creates SSE connection with audio mode parameter
- **Connection Handlers**:
- **onopen** - Enables send button and form submission when connected
- **onmessage** - Processes incoming messages from agent
- **onerror** - Handles disconnections with auto-reconnect after 5 seconds

Message Processing (`app.js`)

Handles different message types from server:

```
  // Handle incoming messages
  eventSource.onmessage = function (event) {
    // Parse the incoming message
    const message_from_server = JSON.parse(event.data);
    console.log("[AGENT TO CLIENT] ", message_from_server);

    // Check if the turn is complete
    // if turn complete, add new message
    if (
      message_from_server.turn_complete &&
      message_from_server.turn_complete == true
    ) {
      currentMessageId = null;
      return;
    }

    // If it's audio, play it
    if (message_from_server.mime_type == "audio/pcm" && audioPla

audioPlayerNode.port.postMessage(base64ToArray(message_from_serv
    }

    // If it's a text, print it
    if (message_from_server.mime_type == "text/plain") {
```

```
      // add a new message for a new turn
      if (currentMessageId == null) {
        currentMessageId = Math.random().toString(36).substring(
        const message = document.createElement("p");
        message.id = currentMessageId;
        // Append the message element to the messagesDiv
        messagesDiv.appendChild(message);
      }

      // Add message text to the existing message element
      const message = document.getElementById(currentMessageId);
      message.textContent += message_from_server.data;

      // Scroll down to the bottom of the messagesDiv
      messagesDiv.scrollTop = messagesDiv.scrollHeight;
    }
```

- **Turn Management** - Detects `turn_complete` to reset message state
- **Audio Playback** - Decodes Base64 PCM data and sends to audio worklet
- **Text Display** - Creates new message elements and appends partial text updates for real-time typing effect

Message Sending (`app.js`)

sendMessage() function sends data to server:

```
async function sendMessage(message) {
  try {
    const response = await fetch(send_url, {
      method: 'POST',
      headers: {
        'Content-Type': 'application/json',
      },
      body: JSON.stringify(message)
    });

    if (!response.ok) {
      console.error('Failed to send message:',
response.statusText);
    }
  } catch (error) {
    console.error('Error sending message:', error);
  }
}
```

- **HTTP POST** - Sends JSON payload to `/send/{session_id}` endpoint

- **Error Handling** - Logs failed requests and network errors
- **Message Format** - Standardized `{mime_type, data}` structure

Audio Player (`static/js/audio-player.js`)

startAudioPlayerWorklet() function:

- **AudioContext Setup** - Creates context with 24kHz sample rate for playback
- **Worklet Loading** - Loads PCM player processor for audio handling
- **Audio Pipeline** - Connects worklet node to audio destination (speakers)

Audio Recorder (`static/js/audio-recorder.js`)

startAudioRecorderWorklet() function:

- **AudioContext Setup** - Creates context with 16kHz sample rate for recording
- **Microphone Access** - Requests user media permissions for audio input
- **Audio Processing** - Connects microphone to recorder worklet
- **Data Conversion** - Converts Float32 samples to 16-bit PCM format

Audio Worklet Processors

PCM Player Processor (`static/js/pcm-player-processor.js`)

PCMPlayerProcessor class handles audio playback:

- **Ring Buffer** - Circular buffer for 180 seconds of 24kHz audio
- **Data Ingestion** - Converts Int16 to Float32 and stores in buffer
- **Playback Loop** - Continuously reads from buffer to output channels
- **Overflow Handling** - Overwrites oldest samples when buffer is full

PCM Recorder Processor (`static/js/pcm-recorder-processor.js`)

PCMProcessor class captures microphone input:

- **Audio Input** - Processes incoming audio frames

- **Data Transfer** - Copies Float32 samples and posts to main thread via message port

Mode Switching:

- **Audio Activation** - "Start Audio" button enables microphone and reconnects SSE with audio flag
- **Seamless Transition** - Closes existing connection and establishes new audio-enabled session

The client architecture enables seamless real-time communication with both text and audio modalities, using modern web APIs for professional-grade audio processing.

Summary

This application demonstrates a complete real-time AI agent system with the following key features:

Architecture Highlights: - **Real-time**: Streaming responses with partial text updates and continuous audio - **Robust**: Comprehensive error handling and automatic recovery mechanisms - **Modern**: Uses latest web standards (AudioWorklet, SSE, ES6 modules)

The system provides a foundation for building sophisticated AI applications that require real-time interaction, web search capabilities, and multimedia communication.

Next steps for production

To deploy this system in a production environment, consider implementing the following improvements:

Security

- **Authentication**: Replace random session IDs with proper user authentication
- **API Key Security**: Use environment variables or secret management services
- **HTTPS**: Enforce TLS encryption for all communications

- **Rate Limiting**: Prevent abuse and control API costs

Scalability

- **Persistent Storage**: Replace in-memory sessions with a persistent session
- **Load Balancing**: Support multiple server instances with shared session state
- **Audio Optimization**: Implement compression to reduce bandwidth usage

Monitoring

- **Error Tracking**: Monitor and alert on system failures
- **API Cost Monitoring**: Track Google Search and Gemini usage to prevent budget overruns
- **Performance Metrics**: Monitor response times and audio latency

Infrastructure

- **Containerization**: Package with Docker for consistent deployments with Cloud Run or Agent Engine
- **Health Checks**: Implement endpoint monitoring for uptime tracking